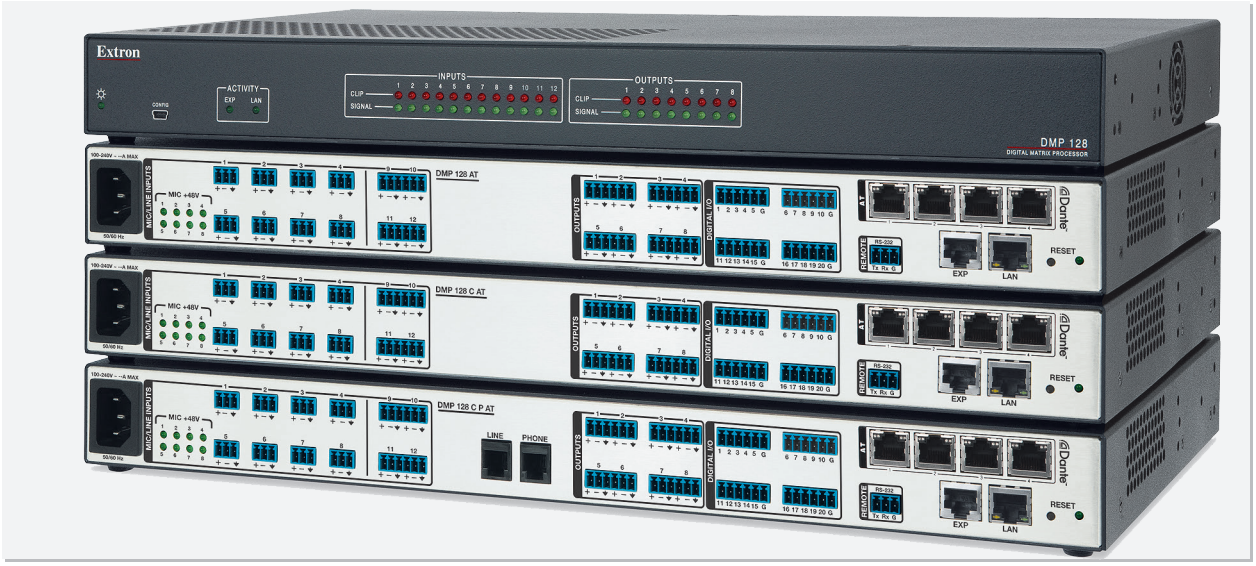
User Guide

Audio Products
Mixers and Processors

DMP 128
Digital Matrix Processor



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68-2036-01 Rev. C
08 13

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Safety Instructions • English

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The following notifications are used in this guide:

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WARNING: A warning indicates a situation that has the **potential** to result in death or severe injury.

CAUTION: A caution indicates a situation that **may** result in minor injury.

ATTENTION: Attention indicates a situation that may damage or destroy the product or associated equipment.

NOTE: A note draws attention to important information.

TIP: A tip provides a suggestion to make working with the application easier.

Software Commands

Commands are written in the fonts shown here:

```
^AR Merge Scene,,Op1 scene 1,1 ^B 51 ^W^C
[Ø1] R ØØØ4 ØØ3ØØ ØØ4ØØ ØØ8ØØ ØØ6ØØ [Ø2] 35 [17] [Ø3]
```

Esc X1 * X17 * X20 * X23 * X21 CE ←

NOTE: For commands and examples of computer or device responses mentioned in this guide, the character "Ø" is used for the number zero and "O" is the capital letter "o."

Computer responses and directory paths that do not have variables are written in the font shown here:

```
Reply from 2Ø8.132.18Ø.48: bytes=32 times=2ms TTL=32
C:\Program Files\Extron
```

Variables are written in slanted form as shown here:

```
ping xxx.xxx.xxx.xxx –t
SOH R Data STX Command ETB ETX
```

Selectable items, such as menu names, menu options, buttons, tabs, and field names are written in the font shown here:

From the **File** menu, select **New**.

Click the **OK** button.

Specifications Availability

Product specifications are available on the Extron website, **www.extron.com**.

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Introduction

This section describes this user guide and the DMP 128, including:

- **About This Guide**
- **About the DMP 128 Digital Matrix Processor**
- **Features**
- **DMP 128 Application Diagram**

About This Guide

This guide contains installation, configuration, and operating information for the Extron Electronics DMP 128 ProDSP Digital Matrix Processor, software controlled digital audio processor.

In this manual, the DMP 128 may also be referred to as “the mixer” or “device.”

About the DMP 128 Digital Matrix Processor

The Extron DMP 128 Digital Matrix Processor is a 12x8 audio mixer featuring Extron ProDSP, automixing, and I/O expansion capabilities, and is available with AEC - acoustic echo cancellation plus Dante™ audio networking. The DMP 128 offers a configuration approach to DSP to simplify mixing, routing, conferencing, and room optimization. Quick and intuitive configuration using the DSP Configurator™ Software allows DMP 128 installation in very little time, with easy-to-learn adjustments heard in real-time. A digital audio expansion port allows two DMP 128 units to be linked together to expand input and output signal management and routing capabilities. The DMP 128 is ideal for presentation and conferencing applications in boardrooms, courtrooms, and conference centers that require advanced matrix mixing with DSP.

The DMP 128 has no front panel controls. All configuration is performed using the Extron DSP Configurator program from a host computer via any of the communication ports; RS-232, USB, or Ethernet (high-speed ports recommended). Signal presence and clip LEDs for the twelve input channels and eight output channels are on the front panel.

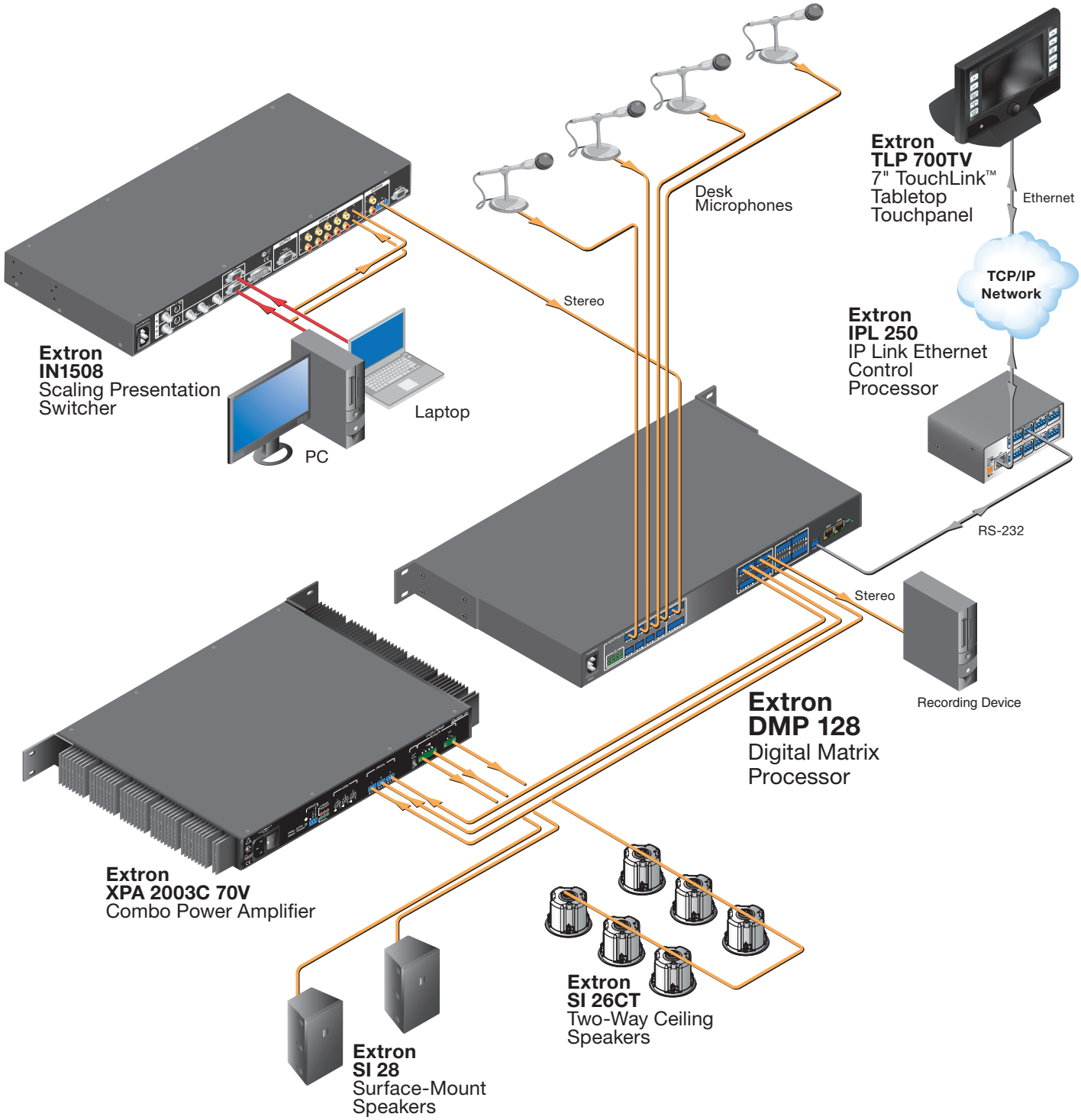
Features

- **Six models with 12 mic/line inputs, 8 outputs and Extron EXP expansion bus include:**
  - DMP 128 — 12x8 ProDSP processor base unit
  - DMP 128 C — 12x8 ProDSP processor with AEC
  - DMP 128 AT — 12x8 ProDSP processor with Dante
  - DMP 128 C AT — 12x8 ProDSP processor with AEC and Dante
  - DMP 128 C P — 12x8 ProDSP processor with AEC and Phone (POTS) interface
  - DMP 128 C P AT — 12x8 ProDSP processor with AEC, Phone (POTS) interface, and Dante

- **Inputs** — Twelve balanced or unbalanced mic/line level on 3.5 mm, 3-pole and 6-pole captive screw connectors, eight with phantom power.
- **Outputs** — Eight balanced or unbalanced line level on 3.5 mm, 6-pole captive screw connectors.
- **Eight channels of acoustic echo cancellation (AEC)** — The DMP 128 C models include eight independent channels of high performance AEC, as well as selectable noise cancellation. Extron AEC features advanced algorithms that deliver fast echo canceler convergence for optimal intelligibility in situations that challenge AEC performance, including double-talk, and the use of wireless microphones at the near end.
- **Dante™ Audio Networking** — Dante equipped DMP 128P AT models provide scalable audio transport over a local area network using standard Internet protocols. Each DMP 128P AT sends out 24 channels of 24-bit/48 kHz digital audio and can receive 56 channels over the network. A built-in four-port Gigabit switch provides direct interconnection of multiple DMP 128P AT units to create larger, cost-effective audio matrixes. Both Dante and the DMP 128P AT processor four-port switch are AVB - Audio Video Bridging ready. Dante is a trademark of Audinate® Pty Ltd.
- **Digital audio expansion port for linking two DMP 128 units** — An expansion port allows any two DMP 128 models to be linked together via a single shielded CAT 6 cable. This allows eight matrix mixes of the inputs, plus eight virtual paths to be sent and received between units.
- **Automixer with eight gate groups** — The DMP 128 features an automixer with advanced features for managing signal levels from multiple microphones. The automixer includes a gating mode that automatically gates channels on or off, as well as a gain sharing mode that maintains the overall system gain based on the number of active mics.
- **ProDSP audio signal processing** — The DMP 128 features 32/64-bit floating point audio DSP processing, which maintains very wide dynamic range and audio signal transparency, to simplify management of gain staging while reducing the possibility of DSP signal clipping.
- **48-volt phantom power** — The DMP 128 is equipped with selectable 48-volt phantom power for the first eight inputs, allowing the use of condenser microphones.
- **24-bit/48 kHz analog-to-digital and digital-to-analog converters** — Fully preserve the integrity of the original audio signal.
- **Fixed, low latency DSP processing** — Input to output latency is low within the DMP 128 and stays constant, regardless of the number of active channels or processes. While latency increases marginally on channels with AEC enabled, overall latency remains low. Fixed latency processing keeps audio in sync with video, and prevents distractions to presenters or performers resulting from delayed live audio.
- **DSP Configurator Software** — A powerful yet user-friendly PC-based software tool for managing all audio operations of the DMP 128. It enables complete setup and configuration of digital audio processing tools on the ProDSP platform, as well as routing and mixing.
- **Intuitive Graphical User Environment** — The DSP Configurator Software features a graphical user environment that offers a clear view of all input and outputs, audio processing blocks, routing, mix-points, and virtual routing in a single screen. This allows a designer or installer to quickly view an audio configuration without having to access multiple dialog boxes or menus.

- **Device Manager** — Device Manager in the DSP Configurator Software enables easy configuration of multiple Extron DSP products, including two linked DMP 128 processors, by toggling between graphical user environments for each unit. Processors can be grouped into folders for organizing as separate rooms or buildings. Settings for multiple Extron DSP products in the Device Manager can be saved to a single file.
- **Flexible control options** — The DMP 128 can be controlled using the DSP Configurator Software and a PC connection to the Ethernet port, the RS-232 serial port, or the USB 2.0 port on the front panel. The DMP 128 can also be controlled through a control system with Extron Simple Instruction Set (SIS)™ commands, and by accessing the internal Web pages.
- **Copy and paste for processing blocks** — To help speed audio system design and setup, parameter settings can be quickly copied between individual processing blocks or identical groups of blocks within the graphical user environment, using conventional cut-and-paste commands.
- **Building Blocks processor settings** — A collection of pre-designed processor settings optimized for a specific type of input or output device, such as microphones and Extron speakers, with preset levels, filters, dynamics, and more. Flexible building blocks are available on each I/O strip and allow system designers to fully customize and save their own building blocks, further streamlining audio system design and integration.
- **Live and Emulate operation modes with configuration file saving** — Live mode allows integrators to connect to the DMP 128 and make live parameter adjustments while hearing or metering them in real-time. This avoids the need to compile and upload a configuration file to the DSP. Emulation mode allows settings to be configured offline, then uploaded to the DMP 128. The software also downloads configuration files from the mixer for archiving. Settings for two DMP 128 processors linked together can be saved to a single configuration file.
- **32 DSP Configurator presets** — Using the DSP Configurator Software, any parameters for DSP processing, levels, or audio routing can be saved as presets. These settings can be saved for the entire system, or any selected group of inputs, outputs, mix-points, and DSP blocks.
- **20 digital I/O ports for remote control or feedback** — Twenty configurable digital I/O ports are provided, so that the DMP 128 can be programmed to sense and then respond to external triggers such as mic activation, muting, and recall of presets.
- **Triple matrix design provides output, virtual, and expansion routing options** — Employs a triple matrix design that offers substantial flexibility in routing, mixing, and processing audio input sources. An output matrix allows any of the twelve inputs to be mixed to any or all eight outputs. If desired, any of the inputs can first be directed into a virtual matrix, which routes the inputs to eight virtual buses, before being mixed back into the output matrix. Virtual buses allow inputs to be processed together as a group. When two DMP 128 processors are linked together via the expansion ports over shielded CAT 6 cable, inputs and virtual buses of one unit can be routed to the other processor through an expansion matrix, for additional processing or matrix mixing into the outputs.
- **Group masters** — The DMP 128 provides the capability to consolidate gain or mute control throughout the system. Gain or mute controls can be selected and added to a group master, which can then be controlled by a single master fader or mute control. Each group master can have up to 16 members, and up to 32 group masters can be created.

DMP 128 Application Diagram



Installation

This section describes the installation of the DMP 128, including:

- **Mounting the DMP 128**
- **DMP 128 Models**
- **Hardware Configuration**
- **Rear Panel Features and Cabling**
- **USB Configuration Port (Front Panel)**
- **Front Panel Indicators**
- **Reset Actuator and LED**

Mounting the DMP 128

The 1U high, full rack width, 8.5-inch deep DMP 128 Digital Matrix Processor can be:

- Set on a table,
- Mounted on a rack shelf,
- Mounted under a desk or tabletop.

For detailed mounting options and UL rack mounting guidelines, (see **Mounting the DMP 128** on page 165).

DMP 128 Models

There are six models of the DMP 128 available. Each model has a different feature set for various applications.

DMP 128 Model Matrix

The following feature matrix provides a breakdown of the various DMP 128 model variations. Where differences occur in operation, they are noted in the text.

	Model	Description
DMP 128		12x8 ProDSP Processor
DMP 128	C	12x8 ProDSP Processor with AEC
DMP 128	AT	12x8 ProDSP Processor with Dante Interface
DMP 128	C AT	12x8 ProDSP Processor with AEC, EXP Bus, and Dante Interface
DMP 128	C P	12x8 ProDSP Processor with EXP Bus, AEC, and Telephone modem
DMP 128	C P AT	12x8 ProDSP Processor with AEC, Telephone modem, EXP Bus, and Dante Interface

Hardware Configuration

The DMP 128 does not have physical controls for configuration or operation.

The DMP 128 has several front and rear panel operational indicators and a rear panel reset button for hardware resets outlined in the following pages.

Rear Panel Features and Cabling

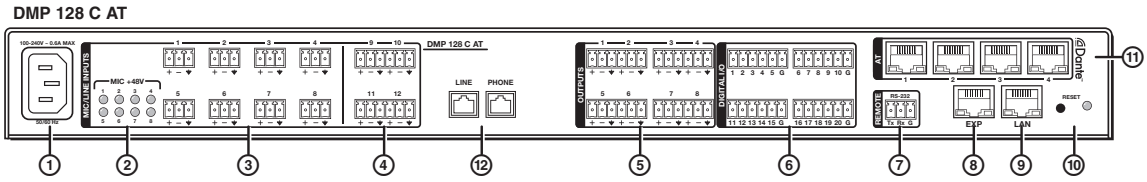


Figure 1. DMP 128 Rear Panel (DMP128 C P AT shown)

① **Power connector** — IEC power connector 100 to 240 VAC, 50 — 60 Hz

② **Phantom Power indicators (MIC +48V)** — LEDs light when +48 V phantom power is placed on the corresponding mic/line input. Phantom power voltage is not adjustable and is only available to Mic inputs 1 through 8.

ATTENTION:

- Condenser microphones require phantom power.
  Dynamic microphones **do not** require power.
  Never set an unbalanced dynamic microphone to +48V. Doing so may damage the microphone.
- For condenser microphones, verify it safely operates at +48 VDC.
- When a line level source is connected, be certain the +48V phantom power is off (cleared).

③ **Mic/Line 1-8 input connectors** — Eight 3-pole 3.5 mm captive screw connectors accept balanced or unbalanced mono mic or line level signals and provide phantom power. Mic/line inputs provide gain settings to accommodate consumer (–10 dBV) and professional (+4 dBu) operating line level sources, plus mic level sources. Up to eight mono mics or line inputs, balanced or unbalanced in any combination can be connected to these inputs, (see figure 2).

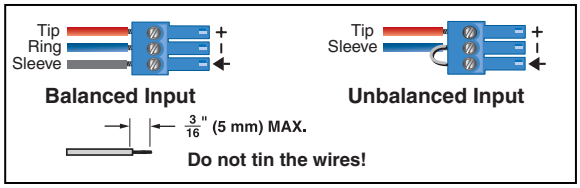
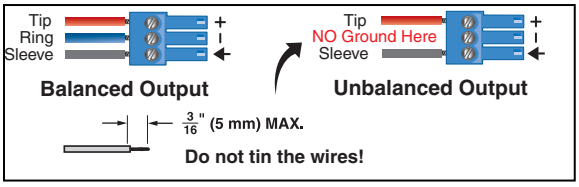


Figure 2. Balanced or Unbalanced Mic and Line Input Wiring

④ **Mic/Line 9-12 input connectors** — Four 6-pole 3.5 mm captive screw connectors accept balanced or unbalanced mono mic or line level signals. Mic/line inputs provide gain settings to accommodate consumer (–10 dBV) and professional (+4 dBu) line level sources, plus mic level sources. Up to four mono mics or line inputs (or two stereo line inputs), balanced or unbalanced in any combination can be connected to these inputs.

⑤ **Mono output connectors** — Four 6-pole 3.5 mm captive screw connectors provide up to eight balanced or unbalanced connections for mono line level output signals.



ATTENTION: Connect the sleeve to ground (⏚). Connecting the sleeve only to a negative (–) terminal will damage the audio output circuits.

Figure 3. Output Connector Wiring

⑥ **Digital I/O output connectors** — Four 6-pole 3.5 mm captive screw connectors each provide five configurable digital input or output ports allowing connection of up to twenty various devices such as motion detectors, alarms, lights, LEDs, buttons, photo (light) sensors, temperature sensors, and other devices.

Digital I/O ports monitor or drive TTL level digital signals. The inputs can be configured to operate in one of two modes: digital input or digital output. In output mode, the device can source up to 250 mA at +5 V. In Input mode, voltages greater than 1 V indicate a logic 'high' signal while voltages less than 1 V indicate a logic 'low'.

All digital I/O ports are tied to a common ground (one common ground for each 6-pole connector), but can be individually configured to operate in one of two modes: digital input or digital output

NOTE: These ports can be configured via the DSP Configurator (see **Digital I/O Ports** on page 88).

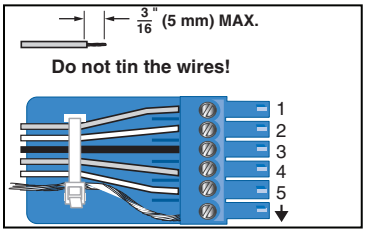


Figure 4. Digital I/O Wiring

⑦ **RS-232 connector** — One 3-pole 3.5 mm captive screw connector, labeled RS-232, for bi-directional RS-232 (±5 V) serial control. Default baud rate is 38400. The RS-232 port is not intended to be used for configuring the DMP 128.

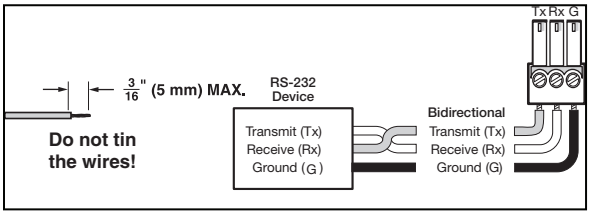
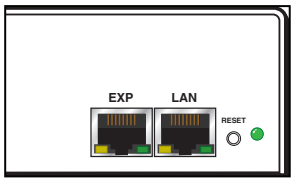


Figure 5. RS-232 Wiring

⑧ **EXP port connector** — One RJ-45 jack for one additional DMP 128 connection. The EXP connector has a green LED to indicate proper connection to an active expansion network and a yellow LED that blinks to indicate data activity.



NOTE: A one foot shielded CAT 6 cable is provided for the EXP connection.

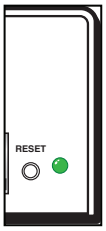
Figure 6. EXP and LAN Connections

⑨ **LAN connector** — A standard RJ-45 jack (see above) accepts a standard Ethernet cable for network connection. The control system and DMP 128 must be connected to the same network.

NOTE: To connect the DMP 128 directly to a computer Ethernet port, use a crossover Ethernet cable.

- A yellow (ACT) LED indicates data activity on the connection.
- A green (Link) LED indicates the jack is connected properly to the network. See **SIS Programming and Control** on page 129 for additional information on Ethernet cabling.

⑩ **Reset button and LED indicator** — The reset button returns the DMP 128 to different tiers of default states and can place the unit into an event recording mode for troubleshooting. When using the reset function, the LED flashes to signify the different tiers (see **DMP 128 Hardware Reset Modes** on page 169). When not in reset mode, the LED operates as a power indicator, duplicating the front panel LED operation.



⑪ **AT connections (AT models only)** — Four RJ-45 jacks for Ethernet connection form a 4-port Gigabit switch that interfaces with the AT bus. The AT port expansion bus uses the Dante protocol for digital media networking allowing connection of multiple DMP 128 AT models to form a larger matrix.

The AT bus supports 56 channels of audio input (Rx) per DMP 128 AT. Output channel support (Tx) includes the eight line outputs, eight virtual returns (post processing), and eight expansion outputs for a total of 24 channels. Audio from an AT port is placed on a network and the audio channels assigned to the network are available to any Dante-compatible device on the network, such as another DMP 128 AT.

NOTE: The Dante Controller software is required for configuration of the AT expansion bus (see **Dante Controller Software Installation** on page 113).

⑫ **Telephone connections (P models only)** — These optional connections provide telephony access.

The POTS interface provides two RJ-11 telephone jacks to connect to the incoming phone line (LINE) and the telephone (PHONE).

The telephone interface follows all applicable US and International standards.

ATTENTION: For telephone and network cabling, to reduce the risk of fire, use a minimum conductor size of 26 AWG, UL Listed or CSA Certified Telecommunication Line Cord.

USB Configuration Port (Front Panel)

A front panel configuration port uses an Extron USB A Male to USB Mini B Male Configuration Cable, **26-654-06** for connection to a PC computer USB port.

The USB 2.0 port uses a mini type-B connector to connect to a host computer for control. The DMP 128 USB driver must be installed prior to using the port (see **Installing the USB Driver** on page 15).

NOTE: The DMP 128 appears as a USB peripheral with bi-directional communication. The USB connection is used for software operation (see **Windows-based Program Control** on page 13), and SIS control (see **SIS Programming and Control** on page 129).

Front Panel Indicators

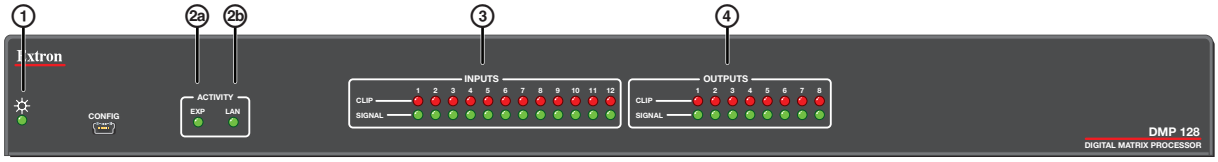


Figure 7. DMP 128 Front Panel

① **Power LED** — The power indicator blinks during power-up and firmware uploads, and lights solid when the DMP 128 is operational.

② **Activity Indicators** — Two green LEDs labeled EXP (2a) for the expansion audio port and LAN (2b) for the standard Ethernet port

2a (non-AT models)

Off — The unit is not connected to a second DMP 128.

On — The unit is connected to another DMP 128 and configured as the primary unit.

Blinking — The unit is connected to another DMP 128 and is currently configured as the secondary unit.

2a (AT Models)

Off — Dante device is not responding.

On — The EXP port is connected to a non-AT DMP 128 and configured as the primary unit.

Blinking — The EXP port is not connected.

2b Indicates activity on the corresponding rear panel Ethernet port connection.

③ **Input Indicators** — Stacked red (signal clipping) and green (signal present) LEDs for inputs 1 through 12 . Each stack represents one input channel.

The green signal LED varies in brightness corresponding to the real-time input or output signal level (see item ④, below). It begins to light at – 60 dBFS increasing in steps to full intensity as the signal level increases. When the signal level reaches – 3 dBFS or above, the red clipping LED lights and remains lit as long as the signal remains above – 3 dBFS. When it falls below that level, the red LED remains lit for 200 milliseconds, after which the display resumes real-time monitoring of the signal level.

④ **Output Indicators** — Stacked red (signal clipping) and green (signal present) LEDs for outputs 1 through 8. Each LED stack represents one output channel.

Reset Actuator and LED

A recessed button on the rear panel initiates several reset modes. The rear panel LED blinks to indicate the reset mode.

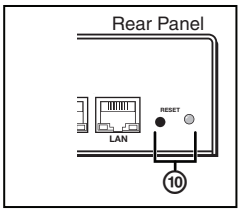


Figure 8. Reset Button and LED

Hardware Reset Modes:

NOTE: The reset modes listed below close all open IP and Telnet connections, and close all sockets.

With power on, when the reset button is held down, the LED blinks every three seconds. At the first blink Mode 3 is available, at the second blink Mode 4 is available, and the third blink indicates Mode 5 is available. The reset modes have separate and distinct functions outlined below (see **DMP 128 Hardware Reset Modes** on page 169).

Mode 1 — Firmware reset: Disconnect power to the DMP 128. Press and hold the reset button while applying power to return the firmware to the version shipped with the unit from the factory. Event scripting does not start when powered on in this mode. This allows recovering a unit with incorrect or corrupt firmware.

All user files and settings are maintained. When returning the unit to an earlier firmware release, some user web pages can work incorrectly.

Mode 3 — Events reset: With power on, press and hold the reset button until the reset LED blinks once (~3 seconds). Release the reset button, then within one (1) second press it again to toggle events on or off, depending on the current state.

- If event logging is currently stopped, following the momentary (<1 sec.) press, the reset LED flashes twice indicating events logging has begun.
- If any events are currently running, following the momentary (<1 sec.) press, the reset LED flashes three times indicating the events logging has stopped.

If the second momentary press does not occur within 1 second, Mode 3 is exited.

Mode 4 — IP Address reset: With power on, press and hold the reset button about 6 seconds until the reset LED blinks twice. Release the reset button, then within one (1) second, press it again to reset the IP settings.

Mode 4:

- Enables ARP program capability
- Sets IP back to factory default IP address (192.168.254.254)
- Sets subnet back to factory default (255.255.0.0)
- Sets gateway back to factory default (0.0.0.0)
- Sets digital I/O port mapping back to factory default
- Turns DHCP off
- Turns events off

If a second momentary press does not occur within 1 second, the reset is ignored.

Mode 5 — Factory default reset: With power on, press and hold the reset button until the reset LED blinks 3 times (~9 seconds). Release then momentarily (<1 second) press the reset button to return the DMP 128 to factory default conditions. If the second momentary press does not occur within one (1) second, the reset is exited.

The default (reset) state of the device is:

- All mix-points are set to 0 dB gain and muted
  - Input 1 is routed to Output 1
  - Input 2 is routed to Output 2
  - Input 3 is routed to Output 3
  - Input 4 is routed to Output 4
  - Input 5 is routed to Output 5
  - Input 6 is routed to Output 6
  - Input 7 is routed to Output 7
  - Input 8 is routed to Output 8
- All outputs active (unmuted, 100% volume).
- No inserted or active DSP processing.
- All audio inputs are set to 0 dB gain and muted.
- All preset and group master memory is clear (empty).

Digital I/O Ports

The four 6-pole 3.5 mm captive screw connector Digital I/O ports provide twenty configurable digital input or output ports designed to connect to various devices such as motion detectors, alarms, lights, LEDs, buttons, photo (light) sensors, temperature sensors, relays (requiring ≥30 mA), and others.

All ports are tied to a common ground (one common ground for each 6-pole connector), but can be individually configured to operate in one of two modes: digital input or digital output.

The ports are configured using DSP Configurator. Each port can be configured to monitor or drive TTL level digital signals (see **Digital I/O Ports** on page 88).

DMP Software

This section describes the control software for the DMP 128, including:

- **Software Control**
- **Windows-based Program Control**
- **DSP Configurator Program Basics**
- **Audio level, Mix-point, Processing Blocks, and Signal Chains**
- **Mic/Line Input Signal Chain Controls**
- **Telephone Rx (DMP 128 C P and DMP 128 C P AT only)**
- **Line Output Channels**
- **Virtual Bus Returns**
- **Output Mix Matrix**
- **Virtual Send Bus Mix Matrix**
- **Expansion Outputs Mix Matrix**
- **Group Masters**
- **Digital I/O Ports**
- **Emulate Mode and Live Mode**
- **Presets**
- **Protected Configuration**
- **DSP Configurator Windows Menus**
- **Optimizing Audio Levels**
- **Signal Path Building Blocks**

Software Control

The DMP 128 can be controlled using the DSP Configurator software, using SIS commands with hyper terminal or DataViewer (see **SIS Programming and Control** on page 129), and accessed using embedded WebPages (see **HTML Operation** on page 155).

The DMP 128 has the following connection options:

- **RS-232** — One single stack 3-pole, 3.5 mm captive screw connector is used for bi-directional RS-232 (±5 V) serial control.
  See **Rear Panel Features and Cabling** on page 6, for additional details on connecting the RS-232 port.
- **LAN** — 10 Mbps, 100 Mbps, half duplex, full duplex connections are supported. Two LEDs indicate connection and activity status. The device has the following default Ethernet configurations:

IP Address:	192.168.254.254	**Default Gateway:**	0.0.0.0
Subnet Mask:	255.255.0.0	**DHCP:**	OFF

  See **Rear Panel Features and Cabling** on page 6, and **Connection Options** on page 129 for additional details on connecting the LAN.
- **USB 2.0** — A Mini B-type USB connector located on the front panel provides high-speed USB 2.0 connectivity to a host computer, backward compatible to 1.0.

Windows-based Program Control

The DSP Configurator Control Program is compatible with Windows XP, Windows Vista, and Windows 7, and provides remote control of the input gain/attenuation, output volume output adjustment, and other features.

DSP Configurator can control the DMP 128 by any of the three control ports, RS-232, USB, or LAN.

Updates to this program can be downloaded from the Extron website at **www.extron.com**.

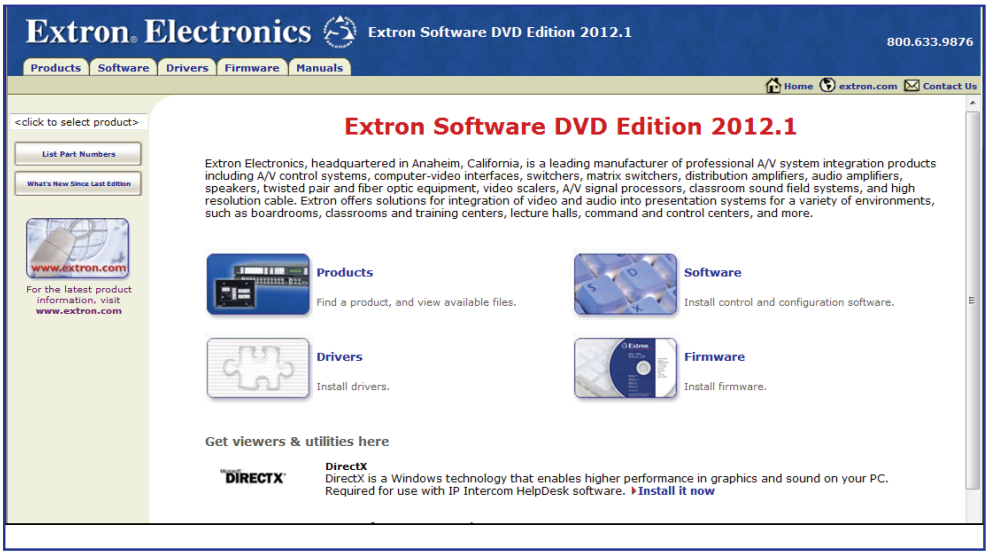
Installing the DSP Configurator Program

The program is contained on the Extron Software Products disk.

Install the software as follows:

1. Insert the disk into the drive.
2. Click the **`Software`** tab or software icon. The software page opens.

> **NOTE:** If the DVD setup program does not start automatically, run `Launch.exe` from the DVD ROM directory using Windows **`My Computer`**.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Extron Electronics, headquartered in Anaheim, California, is a leading manufacturer of professional A/V system integration products including A/V control systems, computer-video interfaces, switchers, matrix switchers, distribution amplifiers, audio amplifiers, speakers, twisted pair and fiber optic equipment, video scalers, A/V signal processors, classroom sound field systems, and high resolution cable. Extron offers solutions for integration of video and audio into presentation systems for a variety of environments, such as boardrooms, classrooms and training centers, lecture halls, command and control centers, and more.
Products
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Install control and configuration software.
Drivers
Install drivers.
Firmware
Install firmware.
Get viewers & utilities here
DIRECTX
DirectX
DirectX is a Windows technology that enables higher performance in graphics and sound on your PC.
Required for use with IP Intercom HelpDesk software. Install it now

Figure 9. DVD Software Menu

3. Scroll to the DSP Configurator program and click **`Install`** to its right.

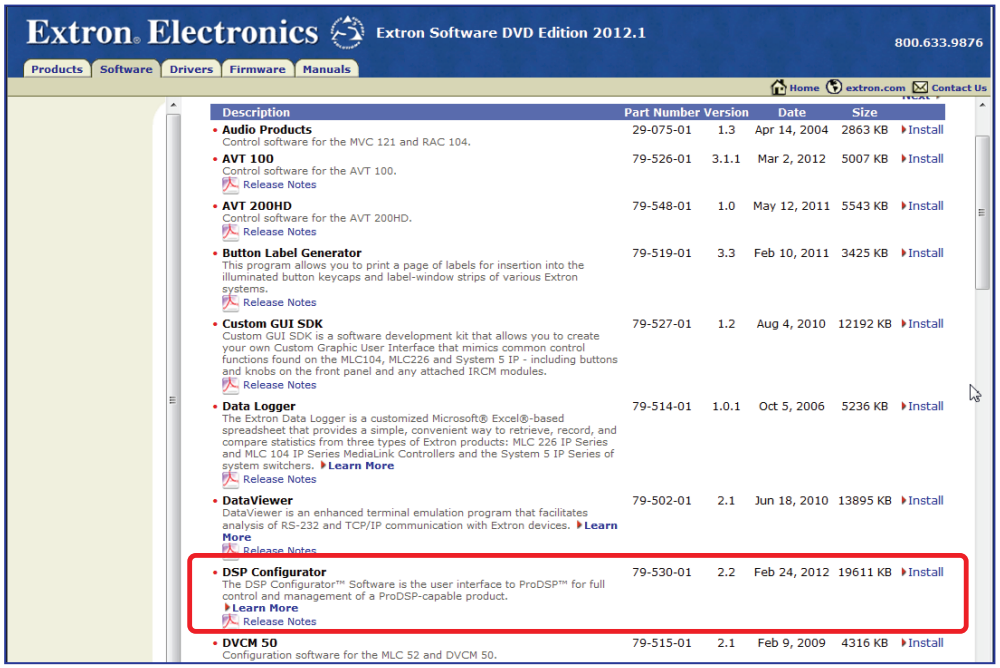

Extron Electronics
Extron Software DVD Edition 2012.1
800.633.9876
Products
Software
Drivers
Firmware
Manuals
Home
extron.com
Contact Us
Description
Part Number
Version
Date
Size
Audio Products
Control software for the MVC 121 and RAC 104.
29-075-01 1.3 Apr 14, 2004 2863 KB Install
AVT 100
Control software for the AVT 100.
Release Notes
79-526-01 3.1.1 Mar 2, 2012 5007 KB Install
AVT 200HD
Control software for the AVT 200HD.
Release Notes
79-548-01 1.0 May 12, 2011 5543 KB Install
Button Label Generator
This program allows you to print a page of labels for insertion into the illuminated button keycaps and label-window strips of various Extron systems.
Release Notes
79-519-01 3.3 Feb 10, 2011 3425 KB Install
Custom GUI SDK
Custom GUI SDK is a software development kit that allows you to create your own Custom Graphic User Interface that mimics common control functions found on the MLC104, MLC226 and System 5 IP - including buttons and knobs on the front panel and any attached IRCM modules.
Release Notes
79-527-01 1.2 Aug 4, 2010 12192 KB Install
Data Logger
The Extron Data Logger is a customized Microsoft® Excel®-based spreadsheet that provides a simple, convenient way to retrieve, record, and compare statistics from three types of Extron products: MLC 226 IP Series and MLC 104 IP Series MediaLink Controllers and the System 5 IP Series of system switchers. Learn More
Release Notes
79-514-01 1.0.1 Oct 5, 2006 5236 KB Install
DataViewer
DataViewer is an enhanced terminal emulation program that facilitates analysis of RS-232 and TCP/IP communication with Extron devices. Learn More
Release Notes
79-502-01 2.1 Jun 18, 2010 13895 KB Install
DSP Configurator
The DSP Configurator™ Software is the user interface to ProDSP™ for full control and management of a ProDSP-capable product.
Learn More
Release Notes
79-530-01 2.2 Feb 24, 2012 19611 KB Install
DVCM 50
Configuration software for the MLC 52 and DVCM 50.
79-515-01 2.1 Feb 9, 2009 4316 KB Install

Figure 10. DVD Control Software Menu

4. Follow the on-screen instructions. By default, the installation creates a `C:\Program Files\Extron\DSP_Configurator` folder for the DSP Configurator program.
5. When the DSP Configurator installation is complete, the USB Installer starts automatically (see figure 11, next page). Extron recommends installing the USB drivers whether they are used immediately or not.

Installing the USB Driver

When the USB installer begins:

1. When the driver installation dialog opens, click **Next** to proceed (a status window tracks the installation).

Extron Electronics - USB Driver Installer v1.0.0 - InstallShield Wizard
Welcome to the Extron Electronics USB Driver Installer
The Extron Electronics USB Driver Installer will install the USB Driver on your computer. To continue, click Next.
WARNING: This program is protected by copyright law and international treaties.
< Back
Next >
Cancel

Figure 11. USB Installer Splash Screen

2. The USB driver installer launches. When the installer completes the installation of the USB drivers, the following dialog opens:

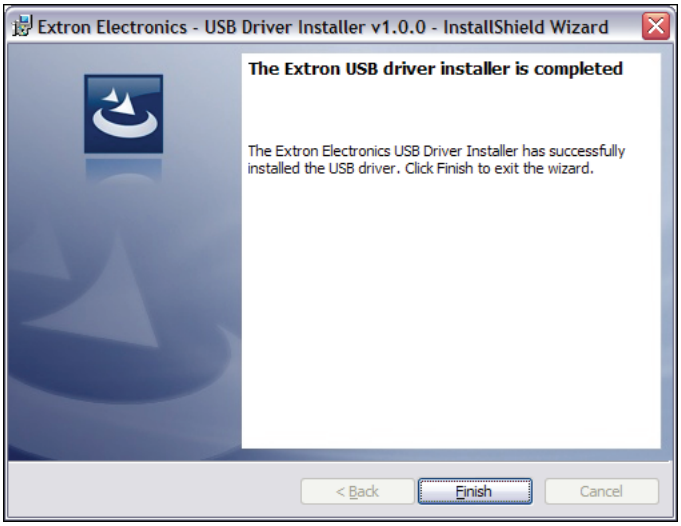

Extron Electronics - USB Driver Installer v1.0.0 - InstallShield Wizard
The Extron USB driver installer is completed
The Extron Electronics USB Driver Installer has successfully installed the USB driver. Click Finish to exit the wizard.
< Back
Finish
Cancel

Figure 12. Successful USB Driver Installation

3. Click **Finish**.

USB driver installation is complete.

DSP Configurator Program Basics

Starting the Program

NOTE: Extron recommends connection via the Ethernet LAN port for running the DSP Configurator program.

To run the DSP Configurator Program, click **Start > Programs > Extron Electronics > DSP Configurator > DSP Configurator.**

DSP Configurator starts in **Emulate** mode (see figure 13, next page). Also see **Emulate Mode and Live Mode** on page 89.

Using the Program

In the DSP Configurator Emulate mode, audio parameters can be selected, then transferred to the DMP 128 by switching to Live mode (while connected to a DMP 128) and pushing the configuration. Audio settings can also be tailored while connected to the DMP 128 for real-time auditioning of the audio output as adjustments are made (see **Emulate Mode and Live Mode** on page 89).

The main screen contains controls for the input and output channels, virtual sends and returns, expansion outputs and inputs, and other information used in the operation of the DMP 128. There is too much information contained on the main screen to enable viewing of the entire mix board at one time, so several methods, outlined on the following pages, are provided to scroll through the screen.

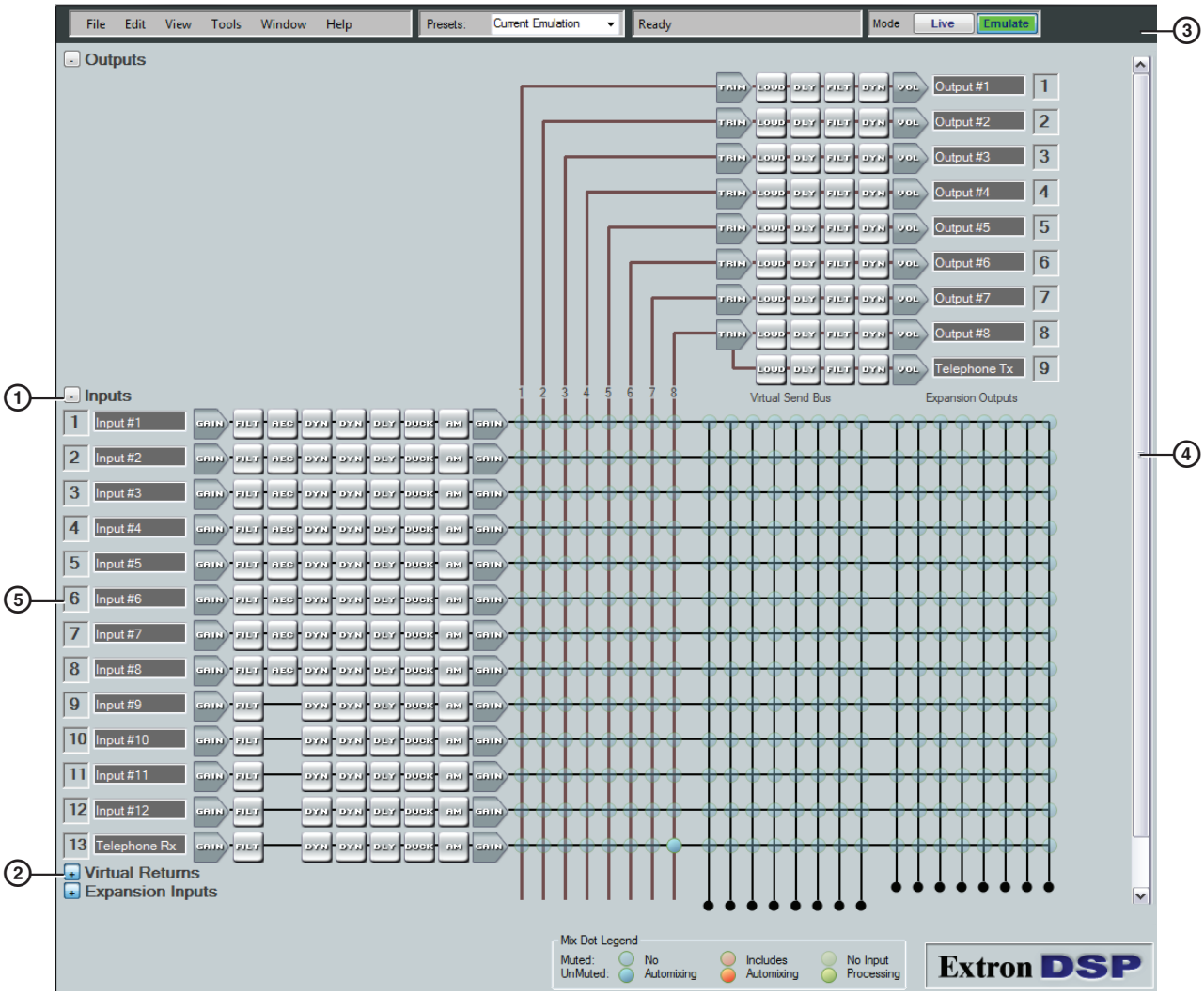


Figure 13. DMP 128 Navigation Aids

① **Minimize buttons** — Click to toggle the view of a selected section from minimum to maximum. For example, the Inputs section is maximized with all processor blocks and the mix-points shown. Clicking the button in this example shrinks the view to its minimum screen area allowing items below to fill the screen.

② **Maximize buttons** — Click to toggle the view of a selected section from maximum to minimum. For example, the Virtual Returns section is minimized with all processor blocks and the mix-points hidden. Clicking the button in this example expands the view to its maximum screen area.

③ **Toolbar** — All tools and functions not directly available on the main screen are found here.

④ **Scroll Bar** — When the sections are maximized such that the screen area takes up more space than can be displayed at one time, items are pushed down or up and no longer appear. Use the scroll bar to bring those items back into view.

⑤ **Channel Numbers** — <Right-click> the channel number to hide a channel that has no device connected or is not used in the current configuration.

NOTE: Hidden channels can be shown again using the tools menu and selecting **`View > Show All Channels`**, then unchecking the hidden channels.

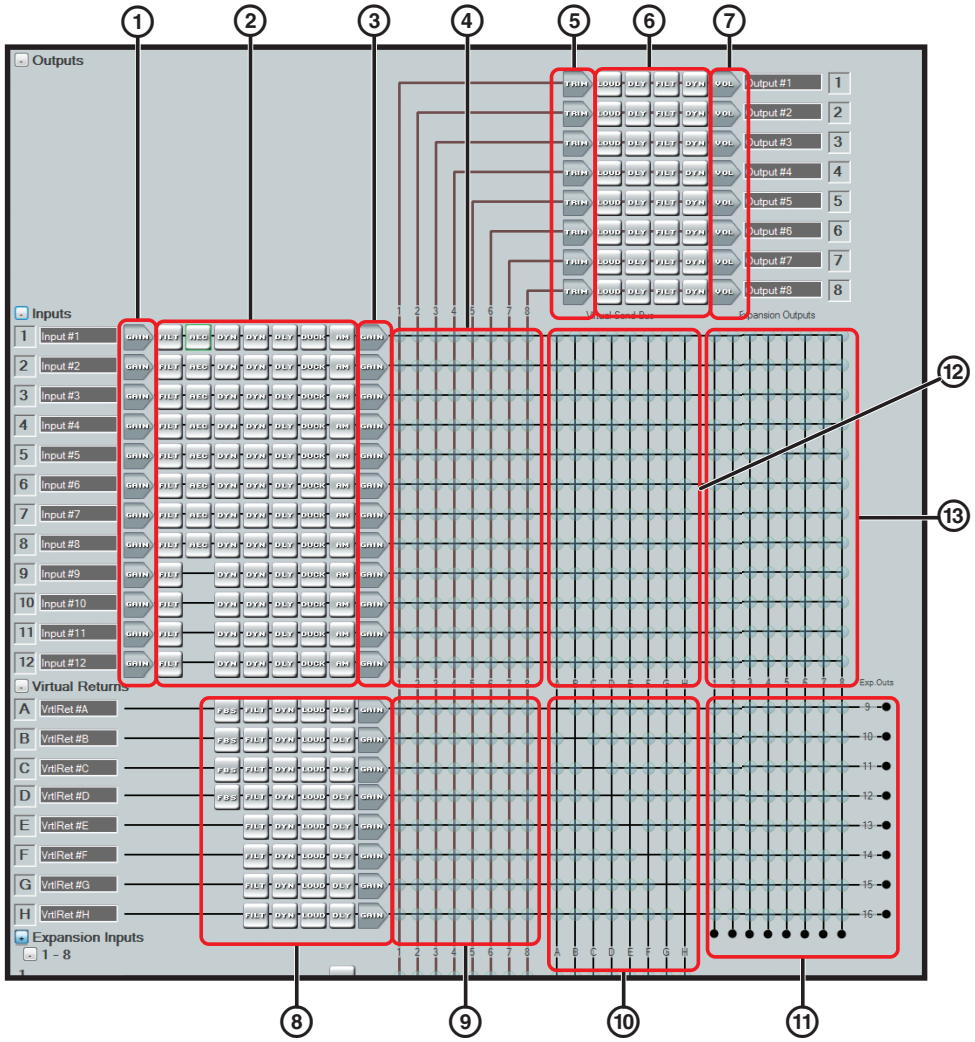


Figure 14. DMP 128 DSP Configurator Main Screen

The DSP Configurator program screen consists of an input and virtual return signal processor chain, the mix-points, and an output signal processing chain.

The main mixer is separated into segments as shown in figure 14.

NOTE: The expansion bus input mix-points are not shown in this view.

① Input gain controls

② Input signal processor channels

③ Input pre-mixer gains

④ Inputs to Outputs mix-points

⑤ Output trim control (post-mixer trim)

⑥ Output signal processor channels

⑦ Output volume controls

⑧ Virtual returns signal processor channels and pre-mixer gains

⑨ Virtual returns to output mix-points

⑩ Virtual returns to virtual sends mix-points

⑪ Virtual returns to EXP sends mix-points

⑫ Virtual send bus to virtual returns mix-points

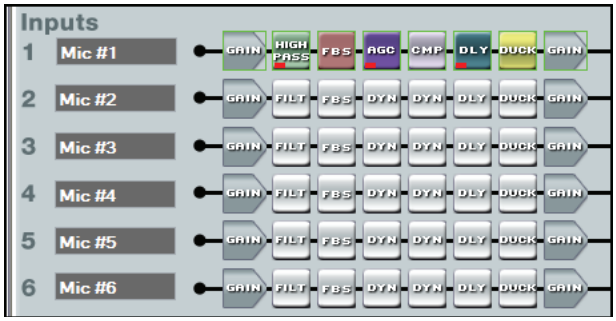
⑬ Inputs to expansion sends mix-points

Cut, Copy, or Paste Functions

The user can cut, copy, or paste a processor. These actions can be performed from a context menu accessed by a right-click on the processor block, using the Edit menu, or using the standard Windows keystrokes: <Ctrl+X> = cut; <Ctrl+C> = copy; <Ctrl+V> = paste.

Multiple elements may be acted upon but the blocks copied must be compatible with the desired paste blocks. A highlighted group of elements can be cut or copied to a clipboard. The clipboard contents can then be pasted, but succeeds if there is an exact one-to-one relationship between the clipboard contents and the block or blocks they are pasted into.

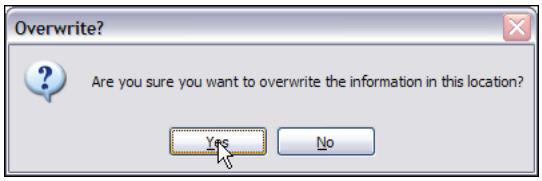
In the following example, the Mic #1 input signal path is copied to Mic #5. First click the mouse and drag it across the entire signal path. The selected blocks are highlighted in green. Press <Ctrl+C>, or use the **Edit > Copy** menu selection to copy the blocks.



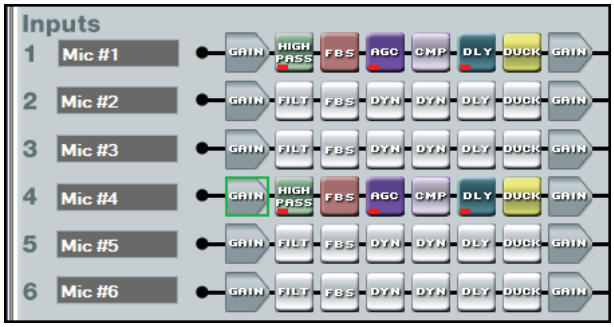
As shown below, the starting point for the paste, (the upper/leftmost element), must first be focused by clicking the mouse on it. Note the green focus outline that appears on the Mic #4 Gain block. The clipboard elements are pasted using the context menu **Paste** command, the **Edit>Paste** command from the toolbar, or <Ctrl+V>.

NOTE: A cut and copy of elements can be pasted to multiple locations. To copy the clipboard to an additional location, click on the leftmost block and paste again.

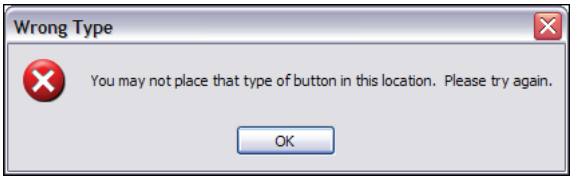
The program warns that all settings in the section pasted to will be overwritten:



Click **Yes**. The entire Mic #4 input path is now identical to the Mic #1 input path including signal levels, parameters settings, and mute/bypass selections.



Any single processor block is copied, then pasted to a similar processor block in the same or different input, virtual or output signal path. Mix-point gains can be copied from one to another. Input gain, pre-mixer gain, post-mixer trim, and output volume can only be copied to like gain blocks. For example, an input gain can be copied to any other input gain, but cannot be copied to a pre-mixer gain, post-mixer trim, or output volume. Mix-point settings can be freely copied between mix-points. The user is always asked whether they want to overwrite the existing information. If an attempt is made to copy a processor block setting to an incompatible block, the user is warned the action cannot be completed.



Navigation

There are two methods of navigation around the screen:

- Keyboard
- Mouse

When a new DSP Configurator file is opened, the upper left element (Output #1 Trim) is the focus by default.

Keyboard Navigation

All screen elements including mix-points have the ability to receive focus using the tab and arrow keys or using the arrow keys following a single click (see **Keyboard Navigation** on page 97).

Mouse Navigation

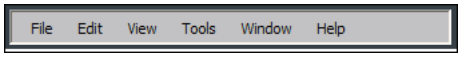
Left-click — Click on a processor block to bring focus to the block, as well as other elements such as tabs, sliders, check boxes associated with the block. Other left-click actions follow the Windows standard. In this user guide “click” always refers to a left-click of the mouse button.

Right-click — A single right-click on a block brings up a context menu specific to that processor block. Other right-click actions follow the Windows standard.

Double-click — Double-click on a box to open it from either the focused or unfocused state of an element.

DSP Configurator Toolbar Menus

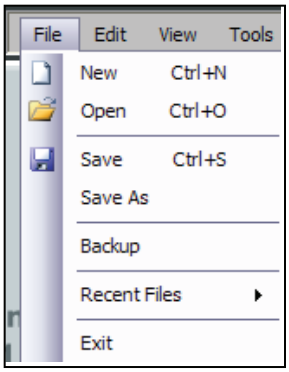
The DSP Configurator contains the following menu bar, arranged horizontally below the title bar:



- **File**
- **Edit**
- **View**
- **Tools**
- **Window**
- **Help**

File

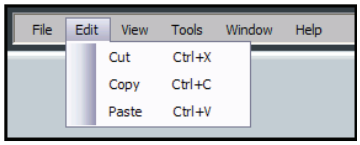
NOTE: **New**, **Open**, and **Recent Files** are unavailable in Live mode.



- **New** — Discards the current DSP configuration (after prompting to save changes) and opens a blank configuration file.
- **Open** — Loads and activates a previously saved DSP configuration file.
- **Save** — Saves all changes to the current DSP configuration file under the current file name. If the file has not previously been saved, prompts for a file name.
- **Save As** — Saves all changes to the current DSP configuration file under a new file name.
- **Backup** — Recalls and transfers all partial presets plus the current configuration to a DSP configuration file within the DSP Configurator program.
- **Recent Files** — Opens a list of recently opened or saved DSP configuration files.
- **Exit** — Closes the DSP Configurator Program.

Edit

- **Cut** — Remove all parameters of a selected processor block or set of selected blocks to the clipboard. If not followed by a **Paste** command to a different block, the parameters are restored.

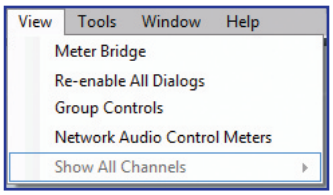


NOTE: Processor blocks are not removed from the processor stream after a **Cut** and a subsequent **Paste** operation. Only the parameters are moved.

Processor blocks and their parameters can be pasted only into another block of the same type. For example, the input 1 filter block and all of its parameters can be copied to the input 2 filter block but not to the input 1 delay block.

- **Copy** — Copies all parameters of a selected processor block, gain block, or set of selected blocks to the clipboard.
- **Paste** — Inserts processor blocks and their parameters from the clipboard into the the location selected.

View



- **`Meter Bridge`** — Opens a Meters dialog box with real-time meters that monitor signal levels at each input and output.

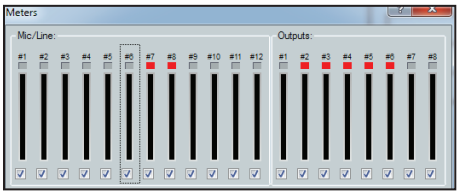


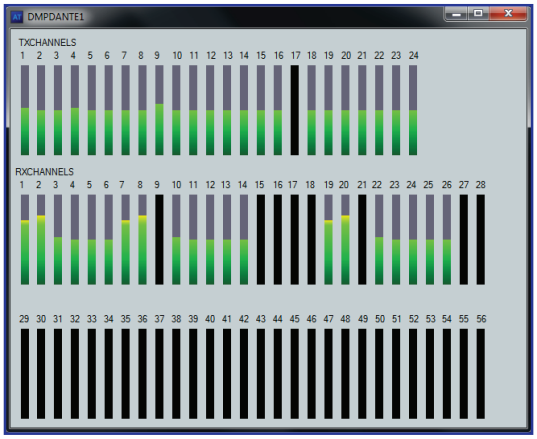
Figure 15. Meter Bridge

> **NOTE:** **`Meter Bridge`** is available in Live mode only while connected using the LAN port.

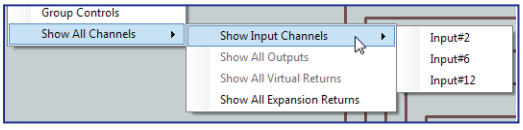
- **`Re-enable All Dialogs`** — Re-enables all dialog boxes, the pop-ups that allow changes to block parameters.
- **`Group Controls`** — Opens the Group Controls dialog box (see **Group Masters** on page 82).
- **`Network Audio Control Meters (AT models only)`** — This menu allows the user to see the AT meters for an attached Dante device.

  To view the meters:

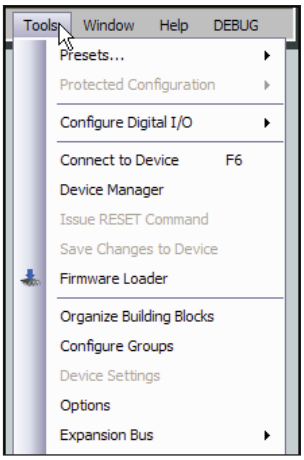
  1. From the main DSP Configurator screen toolbar, select **`View>AT Meters`**.
  2. The `Browse and Select Device` dialog box opens (see right).
  3. Double-click the applicable device. The meter display opens showing the Tx and Rx channels. See **Viewing AT Channels with AT Meters** on page 125 for addtional information.



- **`Show All Channels`** — Enables channels previously hidden from the main menu to be viewed. The selection provides an option to either show all hidden channels for that selection, or by moving to the right, an individual channel can be selected while leaving the others hidden.



Tools



The Tools menu contains the following items and sub-menu:

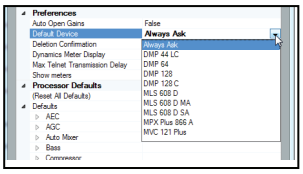
- **Presets** — Provides three options:
  - **Mark All Items** — Mark (select) all parts of the current configuration (excluding presets), including processors and mix-points to save as a partial preset.
  - **Save Preset** — Save the currently marked processors, and mix-points as a partial preset.
  - **Clear Marked Items** — Unmark (deselect) all parts of the current configuration (excluding presets), including processors and mix-points.
- **Protected Configuration (live mode only)**— Allows a user (typically the installer) to save and recall a protected configuration. The protected configuration is useful to place the parameters and values (with the exception of the device IP address) in a known state, either as a troubleshooting tool or as a baseline configuration. The protected configuration, once saved in the device, is always present and cannot be overwritten without entering a user-defined Personal Identification Number (PIN) password. The protected configuration is restored without a PIN.

  NOTE: The default PIN is **0000**.

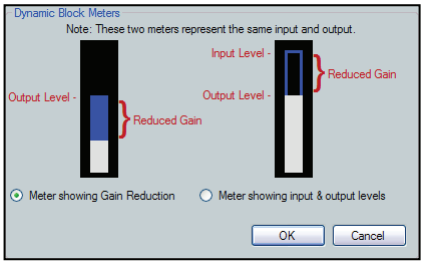
  - **Save** — Save the current configuration (excluding presets), including processors and mixes as a password protected configuration. The DSP Configurator program prompts for a PIN to save.
  - **Recall** — Recall the protected configuration.
  - **Change PIN** — Change the PIN associated with the protected configuration.
- **Configure Digital I/O** — Opens a utility to configure digital I/O ports. The DMP 128 provides twenty digital I/O ports used to trigger external events from internal actions, or for external events to trigger DMP actions (see **Digital I/O Ports** on page 88).
- **Connect to** / **Disconnect from Device (depending on Emulate or Live mode)** — Performs the same functions as the Mode **Emulate** and Mode **Live** buttons.
- **Device Manager** — Opens the Device Manager dialog box. If a device is connected, displays the details (model, MAC address, IP address). In addition, a device can be added or removed, or a selected device cloned, and new folders can be added to an existing device (see **Device Manager** on page 81).
- **Issue RESET Command** — Initializes and clears the following: mix-points, presets, processor blocks, and gain blocks. This reset is identical to the Esc ZXXX↵ SIS command (see **SIS Programming and Control** on page 129).
- **Save Changes to Device (live mode only)** — Saves configuration changes in the DMP 128 to non-volatile memory. This is advised if you are about to power off the device.
- **Firmware Loader** — Launches the Firmware Loader program for firmware updates (see **Firmware Loader** on page 168).
- **Organize Building Blocks** — Provides organization of listed building blocks. You can also import the building blocks file to use your set of building blocks on other computers or export a building blocks file from another computer to use on yours (see **Signal Path Building Blocks** on page 105).

- **Configure Groups** — Opens the configure groups dialog box (see **Group Masters** on page 82).
- **Device Settings (live mode only)** — Opens a dialog box to change the IP address, set administrator and user passwords, change the device name, change the date and time, and to select the serial port baud rate.
- **Network Audio Control** — Launches Dante Controller to facilitate the discovery of networked audio devices that are compliant with the network audio standard used by the DMP 128. Discovery is invoked upon launch, and retrieves device name, audio channels, IP address and the MAC address (see **Dante Controller Configuration** on page 118).
- **Phone Dialer** — The phone dialer utility opens a dialog box that provides telephone service capability for answering and initiating calls to remote attendees participating in a conferencing session (see **Telephone Rx (DMP 128 C P and DMP 128 C P AT only)** on page 58).
- **Options** — Opens a tabbed dialog box that allows customization of the DSP Configurator appearance and operation.

  - **Colors** — Tailor the appearance of the various graphs and dialog boxes. **Appearance** uses a selected color scheme for the complimentary and graph colors. **Complimentary Colors** allows custom selection of colors used with the various graphs and dialog boxes. **Graph colors** change the row colors containing the information and descriptions of the graphs seen in the processor blocks.



  - **Preferences** — The startup splash screen contains options for selection of the devices to connect to, or to **Always ask** on startup. That selection can be changed using **Default Device**.
  - If **Show Meters** is set to **True**, **Dynamic Block Meters** can be used to tailor the appearance of the dynamics meters in order to use the full meter to show input and gain reduction, or to show the level based on the output and gain reduction.

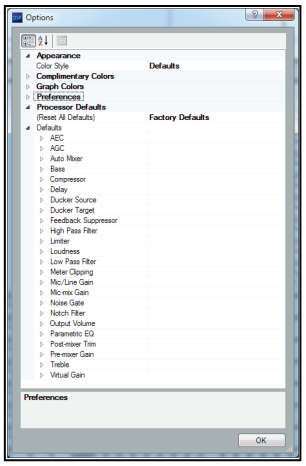


- **Processor Defaults, Reset All Defaults** — Returns the DMP 128 processor and level control blocks to factory default settings. Each processor, and gain/volume/trim block also has an individual default reset.
- **Processor Defaults, Defaults** — Individually selects the default parameters for the various processor, trim, and gain blocks.

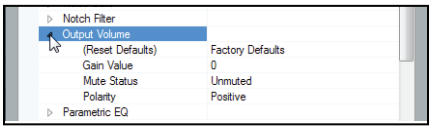
  Each row item contains default settings customized for the processor, filter, trim, or gain block it represents.

  Gain and volume blocks can be initially muted, while filter and dynamics processor blocks can be initially bypassed.

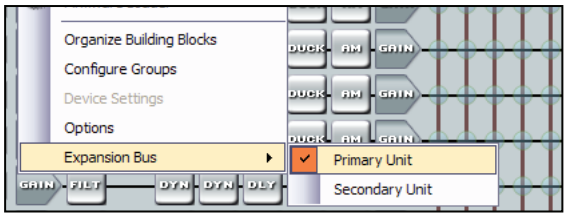
  NOTE: The bypass function is labeled **Enable**.



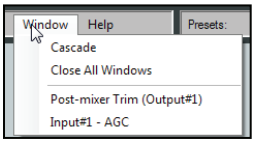
- To view the individual processor defaults, press [►] to the left of the processor, trim, gain, or meter device.



- **Expansion Bus** (live mode only) — Provides a means to select control of either the primary or secondary device (see **Extron EXP Bus** on page 79).



Window Menu



- **Cascade** — Rearranges all open DSP Configurator program screens, including dialog boxes, in a cascading array.
- **Close All Windows** — Closes all open dialog boxes.
- **Individual Windows** — Lists all open dialog boxes. Clicking on the name brings the associated dialog box to the front of the desktop.

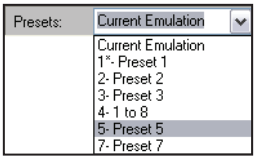
Help Selection

The Help menu contains the following elements:

- **Contents** — opens the Help file at the **Contents** tab.
- **Search** — opens the Help file at the **Search** tab.
- **About...** — displays the name of the application, the current version number, and copyright information.

NOTE: Help can be activated via the F1 key from any main screen or dialog (which accesses context sensitive Help).

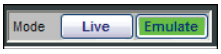
Presets Drop-down



This drop-down list displays up to 32 presets. Select a preset to display and either activate (**Recall**), abort the selection without either recalling or deleting (**Cancel**), or delete it (**Delete**).

NOTE: An asterisk in the drop-down list indicates a preset exists only in the DMP 128 and has not been downloaded to DSP Configurator. After recall, the asterisk is removed.

Mode Buttons



Provides selection between **Live** mode and **Emulate** mode (see **Emulate Mode and Live Mode** on page 89).

Backup

In Live mode (connected to a DMP 128), when presets exist in the DMP 128 that are not present in DSP Configurator (indicated by an asterisk next to the preset name), the function halts and prompts the user to run a backup.

Backup (**File > Backup**) automates the recall of presets from the DMP 128 to a DSP configuration (.edc) file within DSP Configurator, then displays a prompt to save the file to the hard drive. Backup is unavailable when the DSP Configurator program is in Emulate mode.

NOTE: A backup should not be performed during a live event.

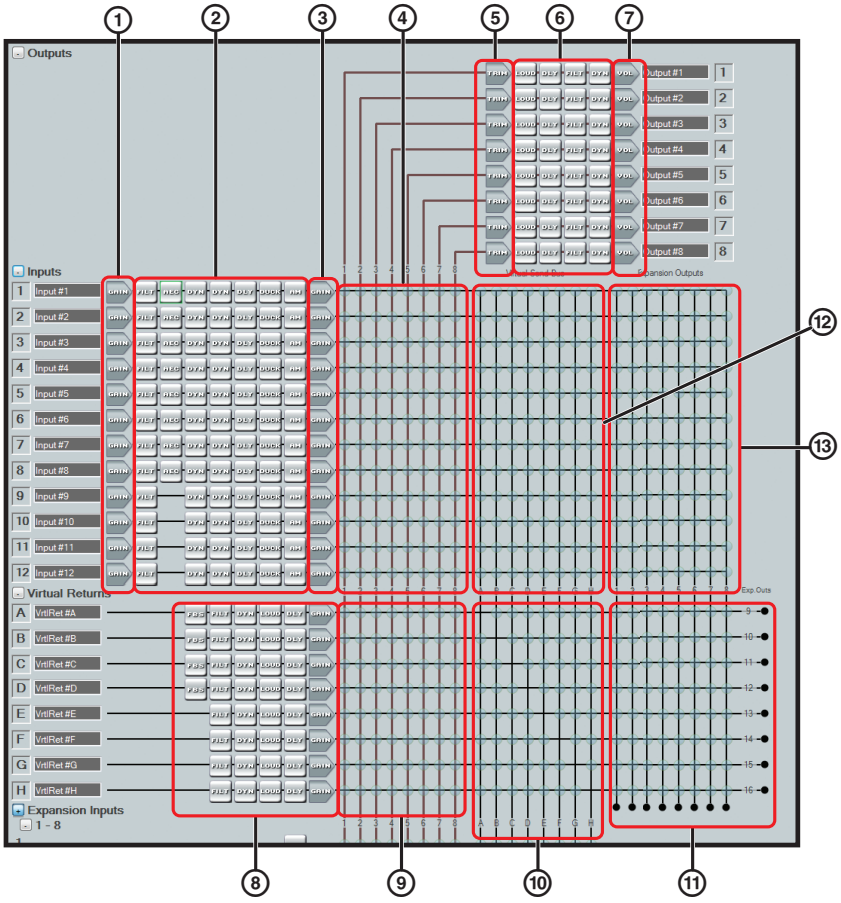


Figure 16. Control Blocks and Processor Chains

Audio Level, Mix-point, Processing Blocks, and Signal Chains

As outlined in red above (see figure 16), all control blocks on the main DSP control screen have one of three main functions in the overall signal chain:

- Level control (gain, trim, volume)
- Mix-point (signal routing)
- Signal processing (filter, AEC, feedback, dynamics, delay, duck, loudness, and automix)

The signal chain varies depending on whether it is in the input, output, virtual bus, or EXP bus stage. Each of the three types of signal processing channels; Input (①, ②, ③), Output (⑤, ⑥, ⑦), and Virtual (⑧) shown in figure 16, consist of a series of two basic types of control blocks specific to that chain: level control (gain ①, ③, trim ⑤, and volume control ⑦), and signal processors (frequency filters, feedback suppression, dynamics, delay, ducking, AEC, AM, and loudness). Both types of control blocks are always present in the chains. By default, gain controls are unmuted and processor blocks are bypassed upon insertion. The default configuration can be modified in the options menu.

The EXP input- bus has only an AM processing block.

Gain, trim, mix-point, and volume blocks can be muted and processor blocks (after being inserted) can be bypassed for signal comparison. Mutes and bypasses are shown by a red indicator in the lower left of the block (see figure 17).

Figure 17. Input Gain Control Muted, Dynamics Processor (AGC) Bypassed

Level Control Blocks

To access a gain, trim or volume control to view a setting, make a change, or observe a live audio meter (input gain and output volume blocks only), double-click the gain block icon (see figure 18). This action opens a dialog box that contains the fader for that control.

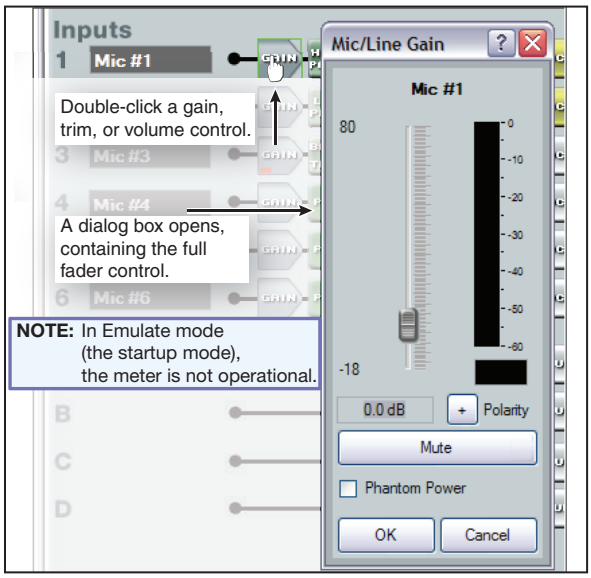


Figure 18. Accessing a Typical Gain Control Dialog Box

Level controls always have a control for setting the signal level and a digital indication of its current setting. They can also have switches or indicators required for their specific function.

Processor Blocks

Each processor block represents a menu of one or more processors that can be inserted into the audio stream. For blocks that provide more than one processor, only one can be selected. Each block can be inserted by a `double-click` or `right-click>Insert` then select the desired processor (see figure 19). Once a block is inserted, the selected processor displays in the block and the block changes color. Processor blocks default to bypassed. Bypass is different from mute since the processor passes an unprocessed signal when in bypass mode. To have them default to "not bypassed", see **Tools** on page 86.

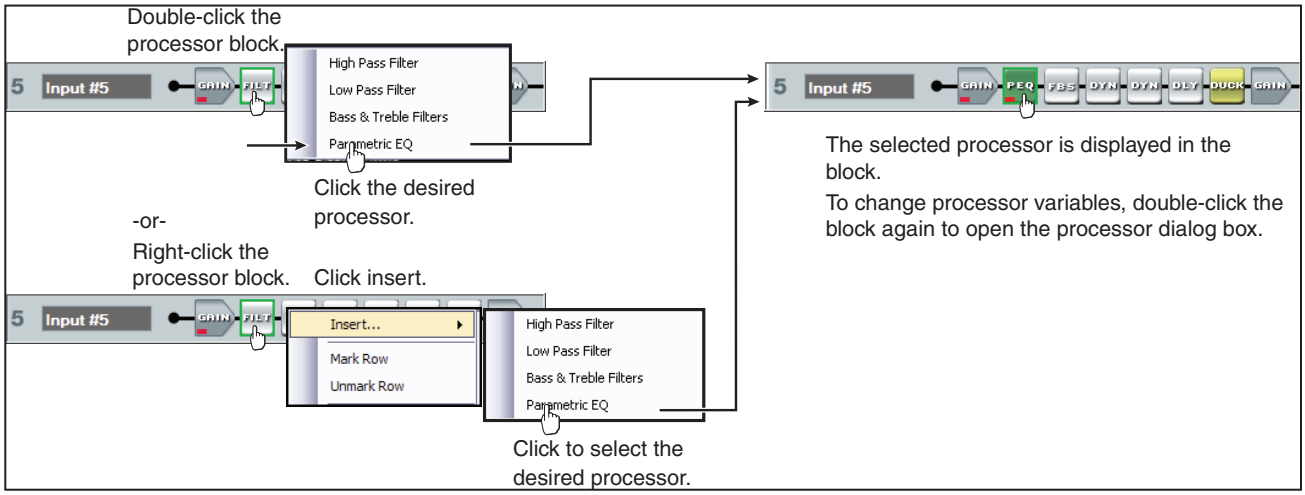


Figure 19. Selecting a Processor Block

Once a processor is inserted, to view associated parameters that define the selected processor (such as a frequency curve) or to remove the bypass, double-click the processor block. This action opens a new dialog box with parameters for the process (see figure 20).

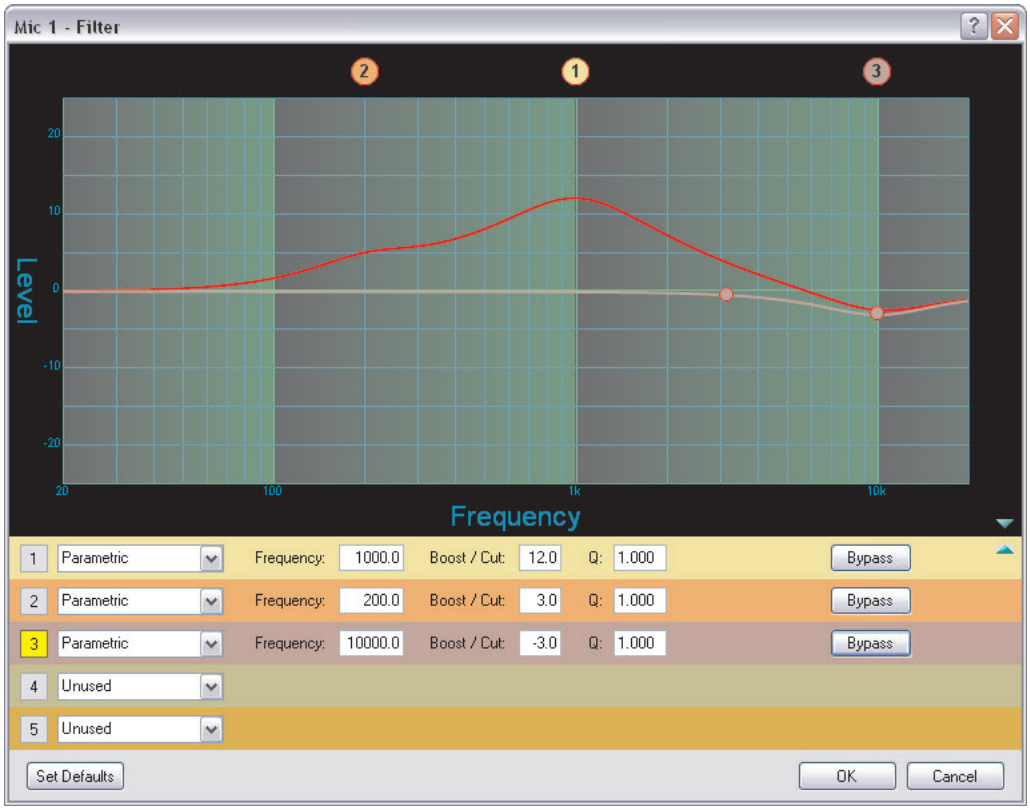


Figure 20. Sample Processor Dialog Box

- Click **Set Defaults** to discard all custom settings and reload the default parameters.

- Click **Bypass** to temporarily suspend processing without removing the processor block. Red indicates the processor is bypassed.

By default, a processor block is bypassed when inserted (the **Bypass** button in the processor dialog box is red).

NOTE: Figure 20 is an example of one type of dialog box. Contents and appearance of each dialog box are unique to the processor type.

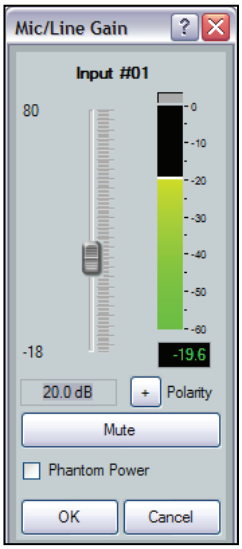
The block can be removed from the signal chain by clicking it to bring focus and pressing <Delete> or by right-clicking and selecting **Delete**.

Detailed explanations of each signal chain with their processor blocks along with mix-point operation follow in the next section.

Mic/Line Input Signal Chain Controls

The input signal processor chain allows adjustments to program or microphone audio material before input to the main mixer.

Gain Control (GAIN)



The gain control provides a single long-throw fader with a range of –18 dB to +80 dB, adjustable in 1 dB increments with the fader or in 0.1 dB increments using direct entry in the level setting readout below the fader. The peak reading meter holds the peak level for one second, displaying it numerically in the box below the meter. The default setting is unity gain (0.0 dB).

The **`Phantom Power`** checkbox, turns the +48 VDC phantom power on (checked) and off (unchecked). Phantom power is typically used to power a condenser microphone.

The **`Mute`** button, silences the mic/line input.

The **`Polarity`** button, allows the polarity of wires connected to the audio connectors (+/tip and –/ring) to be flipped to correct for miswired connectors.

Filter (FILT)

Each filter block allows a total of five filters. The first filter is inserted from a processor list that opens when the block is double-clicked or by selecting it from a context list when the block is right-clicked (see figure 21).

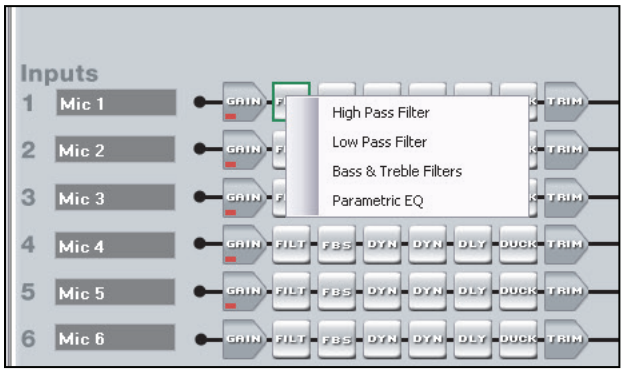


Figure 21. Insert Filter Menu

Once inserted, double-click the processor block to change parameters of the filter. After the first filter is inserted, up to four additional filters can be added to the filter block using the dialog box. Select the desired filters from the following list using the drop-down boxes:

- **High pass filter** — A high pass filter passes a band of frequencies extending from a specified cutoff frequency (greater than zero) up toward the high end of the frequency spectrum. All frequencies above the specified cutoff frequency are allowed to pass, while all frequencies below are attenuated.
  The default cutoff is 100 Hz.

- **Low pass filter** — A low pass filter passes a band of frequencies extending from a specified cutoff frequency (less than infinite) towards the lower end of the frequency spectrum. All frequencies below the specified frequency are allowed to pass, while all frequencies above are attenuated.
  The default cutoff is 10 kHz.
- **Bass and treble filters** — Also known as shelving or tone controls, the separate bass and treble filters cut or boost gain linearly above or below a specific frequency, with the end-band shape giving the visual appearance of a shelf. The bass default frequency is 100 Hz and the treble default is 8 kHz.
- **Parametric equalizer filter** — The parametric filter is a frequency equalizer that offers control of all parameters, including amplitude (the amount of gain/boost or gain reduction/cut applied), center frequency (frequency), and range of affected frequencies (Q) around the center frequency.

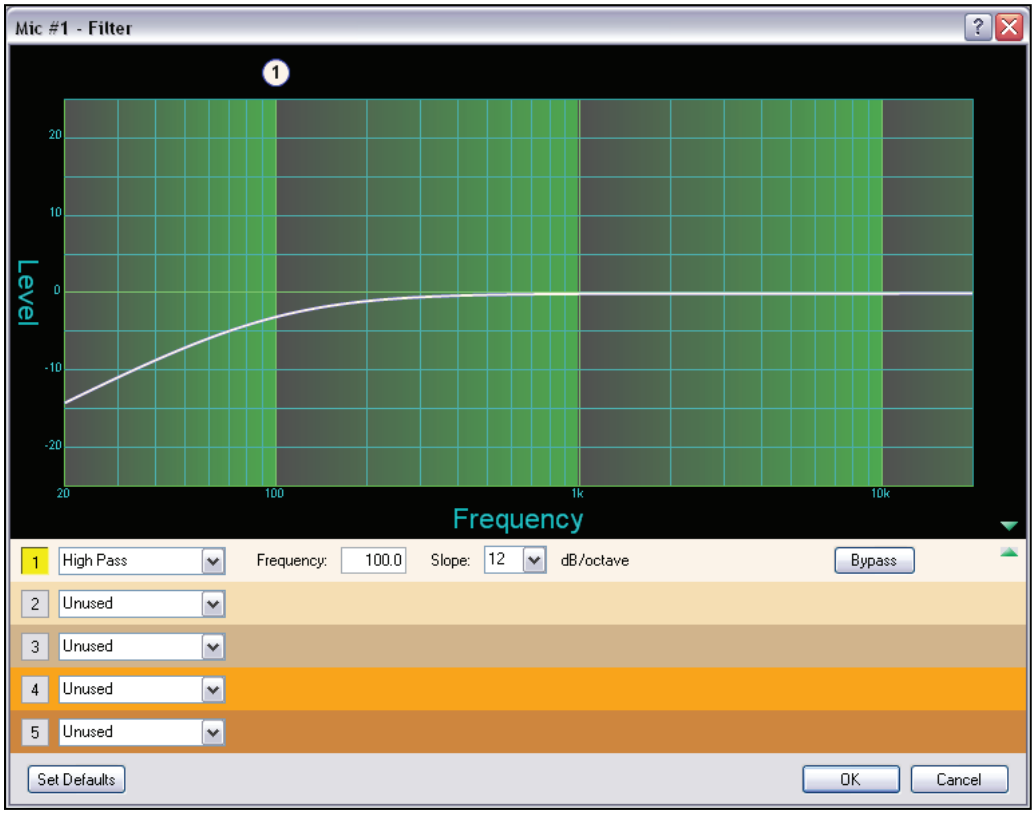


Figure 22. Filter Block Dialog Box

Open the filter block dialog box to insert additional filters. Select a filter type from the drop-down filter selection list. All filter parameters are modified using the Filter block dialog box. Each filter loads with all applicable parameters displayed to the right of each filter selection.

A balloon number appears at the top of the frequency graph display (see figure 22, ①) corresponding to the row number underneath the graph. The location of the balloon number is at the frequency selection of that row.

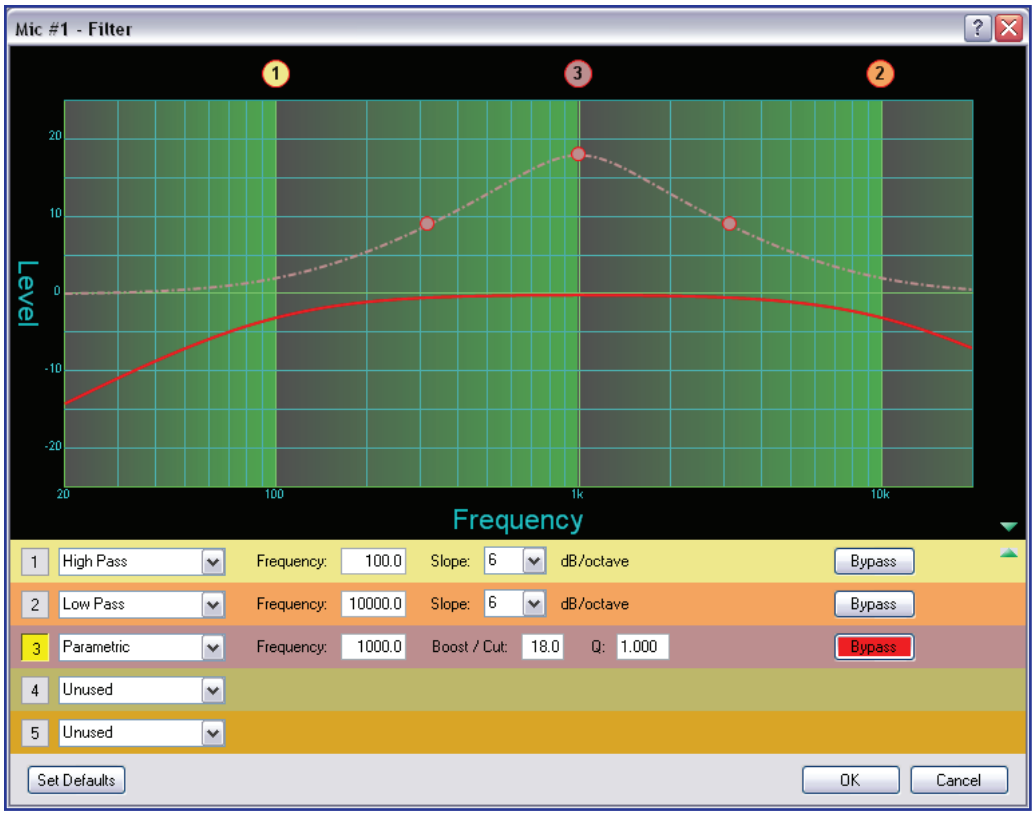


Figure 23. Filter Dialog Box, Filters Added

Within the dialog box, a filter is focused when a filter type is inserted, or is focused by clicking the filter number to the left of the filter selection drop-down list. Note the box number in row 3 in figure 23 is highlighted in yellow, indicating it is the filter in focus. The results of the filter in focus (independent of other filters) show in the graph as a dotted line the same color as its filter row when bypassed. When active (not bypassed), the line is solid.

When multiple filters are enabled, the graph indicates the focused filter result (independent of other filters) in the color of the filter row in table. The composite response, the combined effect of all filters not bypassed, is always displayed in red.

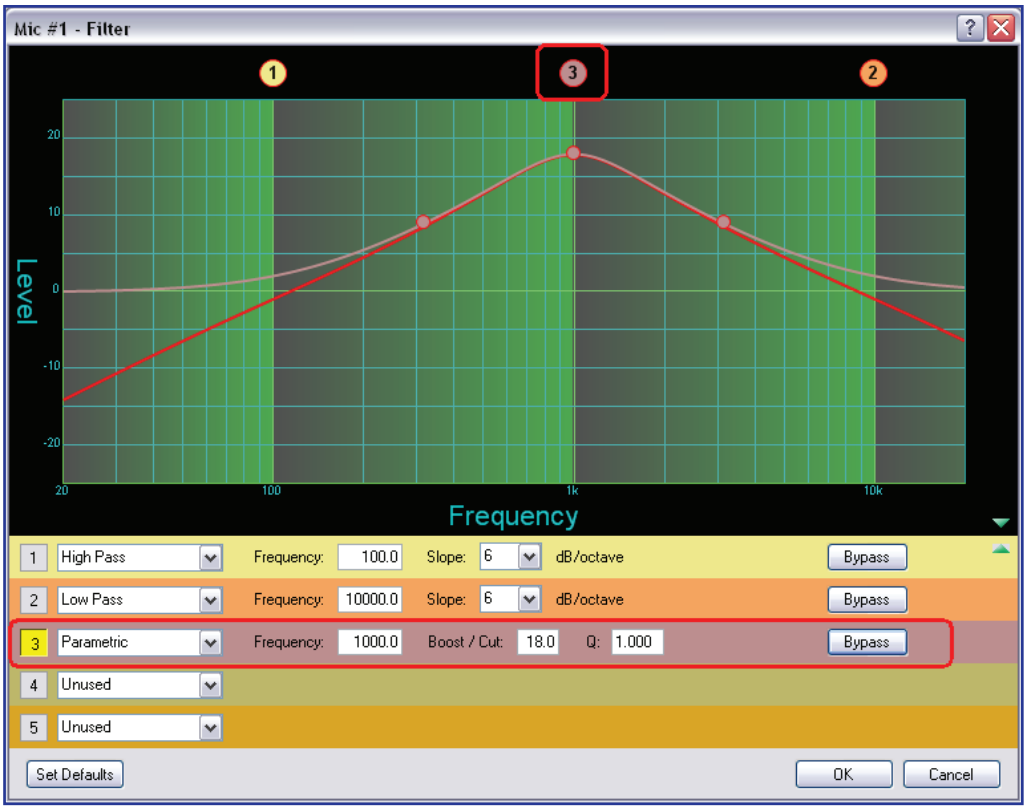


Figure 24. Filter Dialog Box, Filter Not Bypassed

Above the graph, each filter has a "handle" (circled in red above) placed directly above the cutoff or center frequency whose number corresponds to the filter number (outlined in red). Click a handle or click the table row to bring focus to that filter. Click+hold+drag the handle horizontally to change the cutoff or center frequency.

The table below shows each filter type with default parameter settings. The table immediately following shows the possible range for each parameter.

Type	Frequency	Parameter 1	Parameter 2
High Pass	100 Hz	Slope: 6 dB	N/A
Low Pass	10000 Hz	Slope: 6 dB	N/A
Bass	100 Hz	Boost/Cut: 0.0 dB	Slope: 6 dB
Treble	8000 Hz	Boost/Cut: 0.0 dB	Slope: 6 dB
Parametric	1000 Hz	Boost/Cut: 0.0 dB	Q: 1.0

Filter Parameter	Settings Range
Frequency	20 Hz to 20 kHz
Boost/Cut	-24.0 dB to +24.0 dB
Q (Parametric EQ only)	0.707 to 15.000
Slope (HP & LP filters only)	1st Order (6 dB) and 2nd Order (12 dB)

High Pass

The high pass filter allows all frequencies above the specified cutoff frequency to pass unattenuated. All frequencies below the cutoff are attenuated.
The default cutoff is 100 Hz.

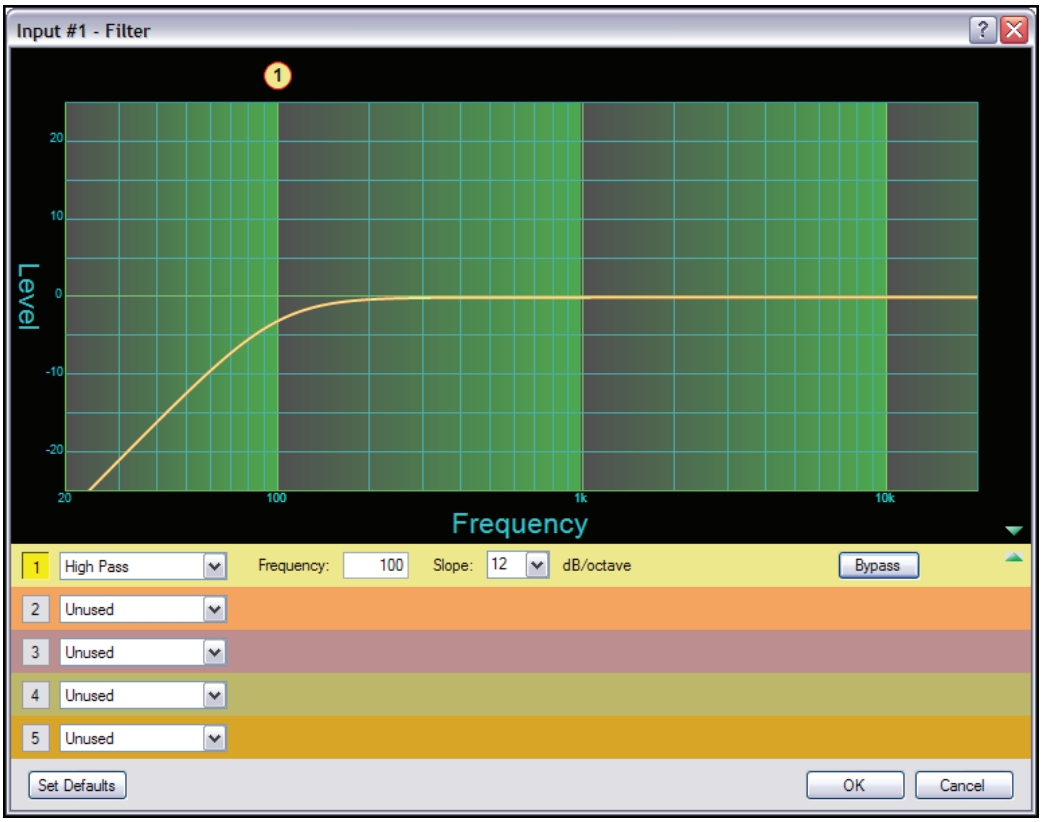


Figure 25. High Pass Filter Response Curve

In figure 25, all frequencies lower than the specified frequency, 100 Hz, are attenuated leaving the upper frequency response flat. Also note at the specified frequency (100 Hz), the signal is about 3 dB down, typical operation for high pass filters.

Low Pass

The low pass filter is the opposite of the high pass filter. All frequencies above the specified frequency are attenuated allowing lower frequencies to pass.
The default cutoff is 10 kHz.

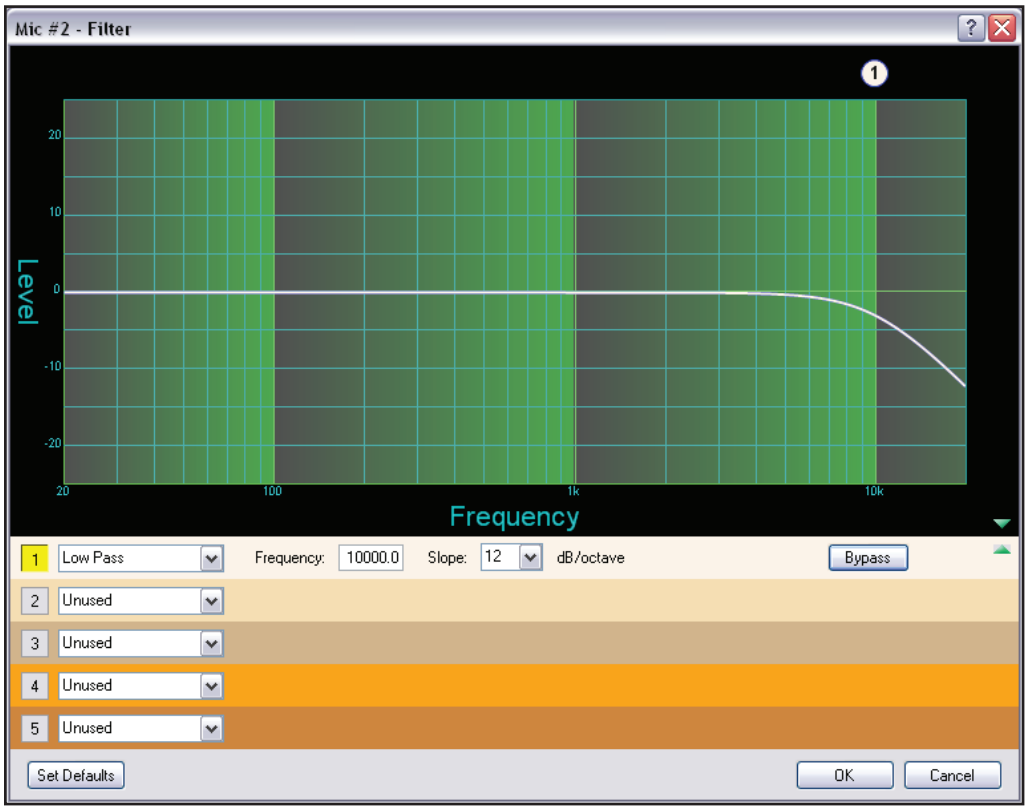


Figure 26. Low Pass Filter Response Curve

Here, the frequencies higher than the specified frequency, 10 kHz, are attenuated leaving the lower frequency response flat.

Bass and Treble Shelving

Bass and treble shelving can be added to the filter. Known as shelving or tone controls, the separate bass and treble filters provide the ability to cut or boost gain linearly above or below a selected frequency, with the end-band shape giving the visual appearance of a shelf.

If only a bass or only a treble filter is required, either bypass the unneeded control or set it to **Unused** in the selection box.

The bass default frequency is 100 Hz and the treble default is 8 kHz.

NOTE: Selecting **Bass & Treble Filters** inserts two separate filters.

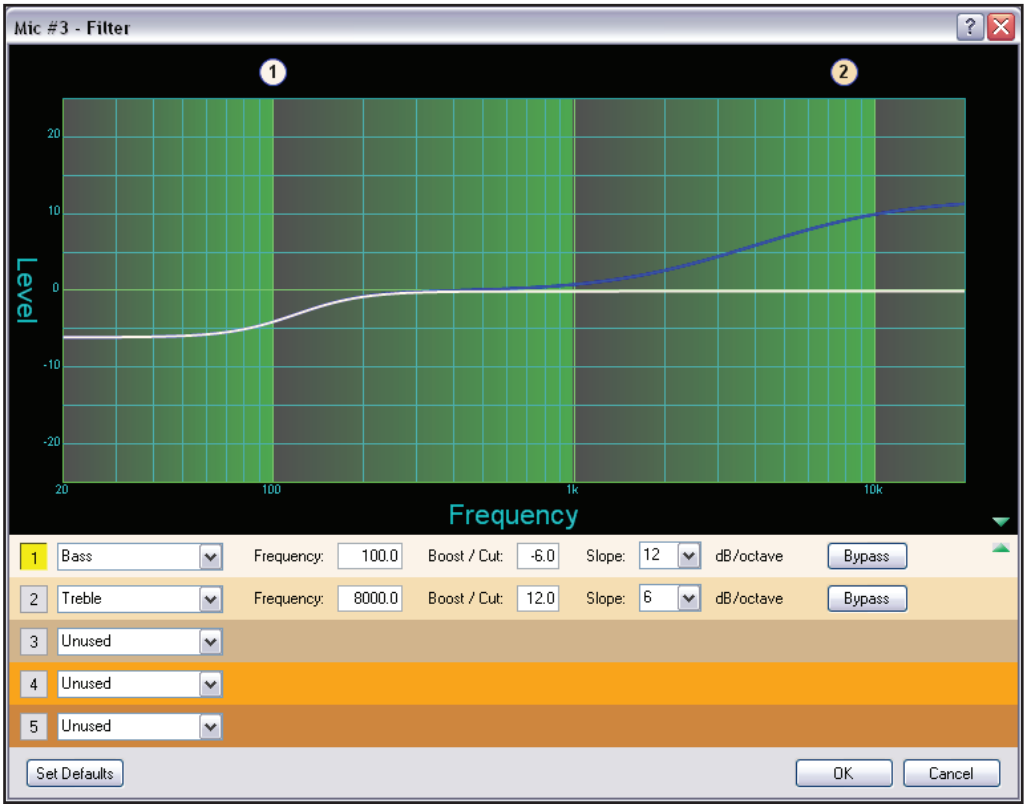


Figure 27. Bass and Treble Shelving

The corner frequency of the controls can be selected to 0.1 Hz accuracy. Two slopes, 6 and 12 dB/octave are available, along with the ability to boost or cut the signal up to 24 dB.

Parametric (Equalizer)

The parametric filter is a frequency equalizer that offers control of all parameters, including amplitude (the amount of gain [boost], or gain reduction [cut] applied), center frequency (frequency), and range of affected frequencies (Q) around the center frequency. Q is the center frequency divided by the bandwidth.

Up to five parametric filters can be placed in the filter block at one time, each set to a different frequency creating a five band parametric equalizer. The control can boost or cut the center frequency, and by changing the Q value, the range of affected frequencies can be widened or narrowed around the center frequency. In general, a higher Q value results in a narrower affected bandwidth.

To demonstrate how Q affects the filter, the following filter block (see figure 28) contains five parametric filters centered at different frequencies, but with the same Q of 1.0. The filter in focus (③) has a center frequency of 1000 Hz boosting that frequency +12 dB over a Q of 1.0. Note the markers on either side of the peak frequency are at about 300 Hz on the left and 3000 Hz on the right, a bandwidth of 2700 Hz.

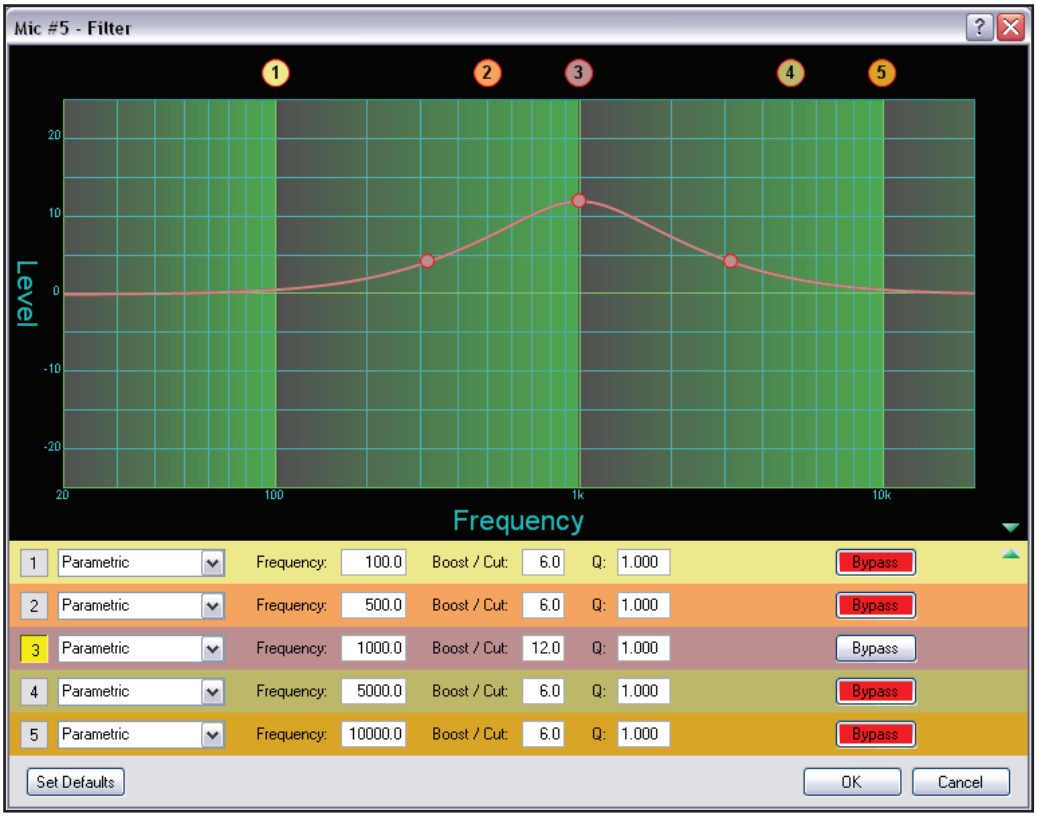


Figure 28. Parametric Filter at 1000 Hz, Q: 1.000

By increasing the Q to 10.000, the center frequency remains the same. The markers show the bandwidth of the filter narrowed to between 900 Hz and 1200 Hz, or about 300 Hz (see figure 29). Using the Q value, parametric filters can be used to notch out a very narrow, or very wide range of frequencies.

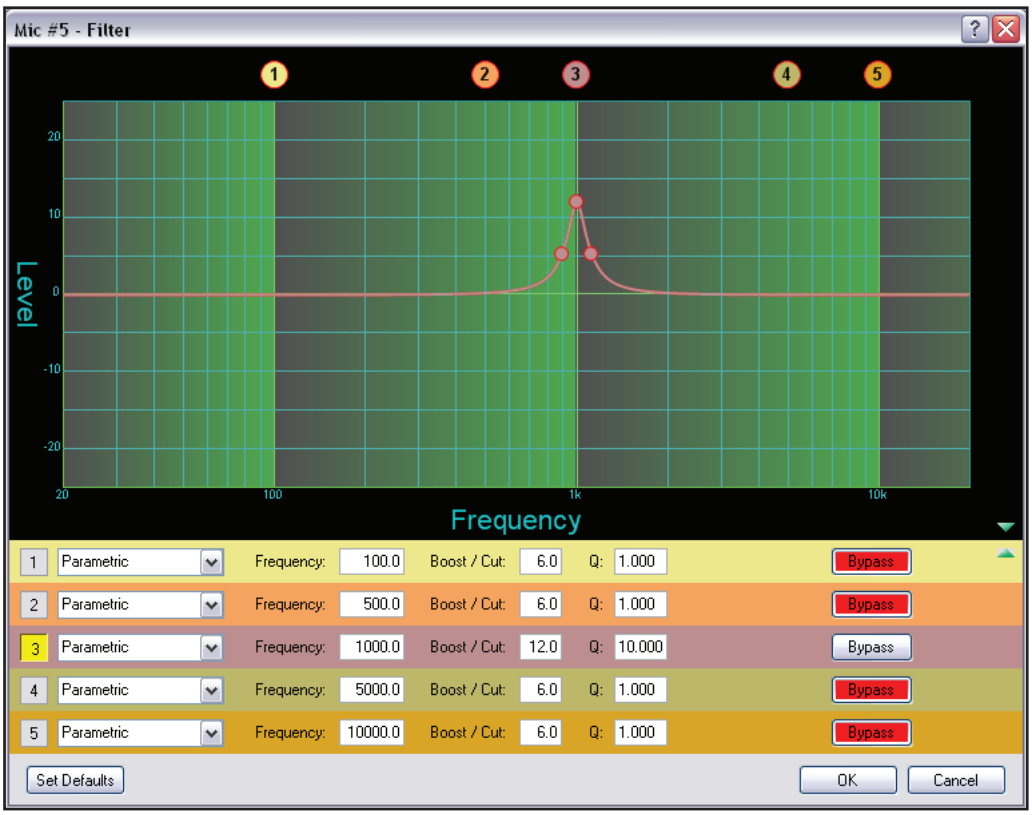


Figure 29. Parametric Filter at 1000 Hz, Q: 10.000

The dialog box above shows the frequency curve for a single active filter. To add its effect to the overall frequency response, remove the bypass on the other filters by clicking **Bypass**.

The overall frequency response is now shown as a solid red line with the filter in focus located in row 3 (see figure 30 below) shown in the color of its table row.

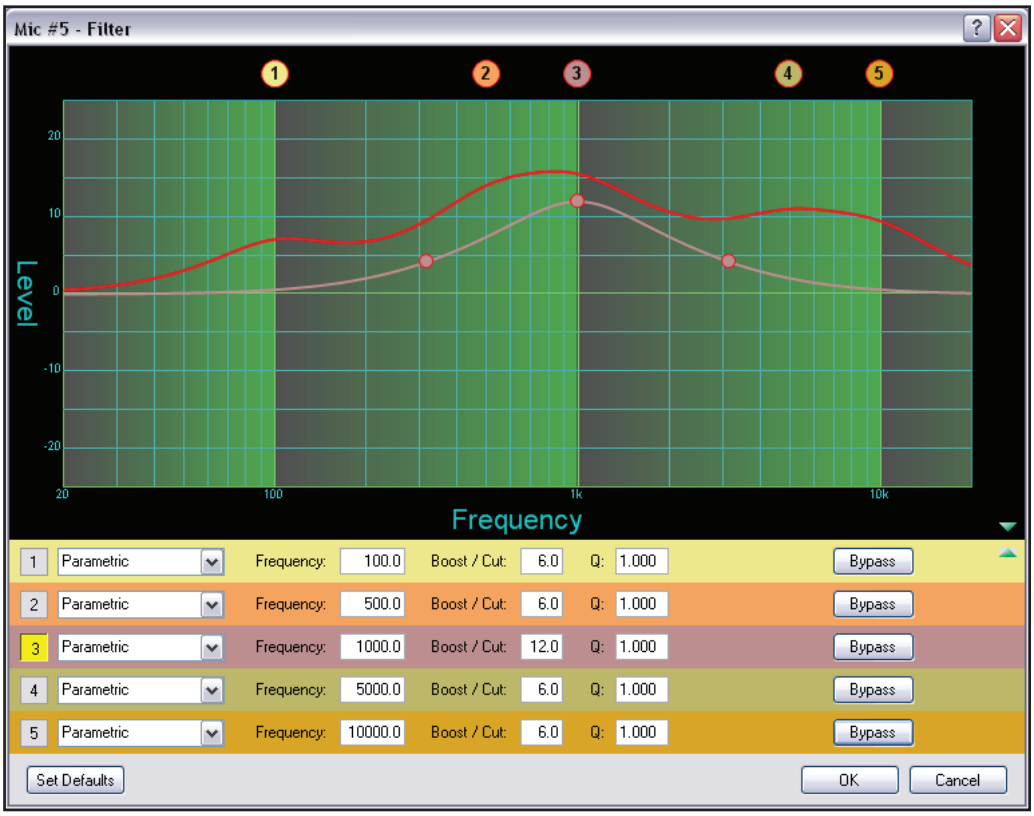


Figure 30. All Parametric Filters Active

The parametric filter allows frequency selection accurate to 0.1 Hz and either 6 or 12 dB of slope.

Acoustic Echo Cancellation (AEC)

The DMP 128 C models provide one acoustic echo canceller processor for each of the first eight mic/line inputs. A single reference can be selected for each AEC from a list of the twelve line inputs.

About AEC

Echo occurs when audio from a talker in the far end is received and amplified into the near end listener's room, with that sound then being picked up by microphones in the near end acoustic space and sent back to the far end. The amount of signal sent back to the far end talker can be substantial, and with the added transmission delay, the result is an echo effect that seriously compromises communication in a teleconference or videoconference.

The Acoustic Echo Cancellation processor removes the potential echo signal at the near end mic channel by comparing it to the received signal from the far end, designated as the "reference," and then creating an adaptive filter to cancel the potential echo before it is sent back to the far end.

AEC Setup

Successful operation of the AEC processing block is a function of proper gain structure and selection of the AEC reference (see **Optimizing Audio Levels** on page 100). This section provides an overview of the two elements.

Proper gain structure involves the relationship between the signal at the selected reference and the signal at the mic input, within the context of proper levels for the reference and mic inputs independently. The mic input gain setting is naturally optimized for the voice level of the talker in that room; therefore the amount of signal from the far end that is picked up by the mic is dependent on how much that far end signal is being amplified in the near end room and the distance from the mic to the speakers.

The reference signal is the signal received from the far end, which is ultimately sent to a sound reinforcement system within the near end room. The output of the video codec might be connected to any of inputs 9 – 12.

In the AEC dialog, a reference can be chosen from any channel in one of three signal chains:

- Input Channels
- Virtual Return Channels
- Output Channels

Extron recommends using an input channel as the reference. An output channel or a virtual channel can also be used as a reference; however, doing so adds a little more delay to the signal being referenced.

Using an output or virtual channel reference allows for the combining of input channels to a single reference, for example, in a conferencing setup where both a telephone and a video codec are used. In this case, both the telephone and video codec input channels can be routed to an output or virtual return, with that output or virtual return then chosen as the reference.

When using an output channel as a reference, the reference point is post volume control; therefore, changes to the listening volume in the room affects the AEC gain structure (see **AEC Dialog**, below). If you have an output channel on the DMP 128 that is not being used, you can isolate the reference channel from the channel being used for volume control by routing reference signals to the unused output channel.

If you do want the reference signal to track with changes in listening volume, and want more control over the actual reference level:

1. Route the far end signal to both the amplifier output and the virtual (unused) output.
2. Create a group master control that contains the amplifier output and virtual output. Set soft limits for the group master control, as desired.
3. Set optimal level for the amplified output. Set optimal level for virtual output. Relative levels between both settings will be maintained by using the group master control.

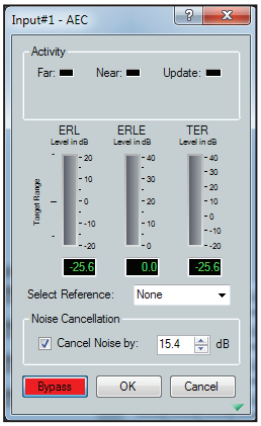
AEC Configuration

To insert and configure an AEC processor:

1. Insert an AEC processor on the desired input channel using one of the following methods:
   - Double-click the AEC (filter) block in the DSP Configurator workspace.
   - Right-click the AEC block to open the context menu and select **`Insert AEC`**.
   - Click the AEC block to select it (or use the arrow keys to navigate to the AEC block) and press <Enter>.
2. Double-click the AEC processor to open it. Open the **`Select Reference`** drop-down list and select a reference.
3. Click **`Bypass`** to disengage bypass. The AEC processor is now operational.

AEC Dialog

The AEC dialog contains a number of meters and indicator LEDs that are essential for setting up gain structure and monitoring activity. The AEC reference must be selected from a list, otherwise the echo canceller will not work. Noise Cancellation, part of the AEC processor, is selected and adjusted here. A detailed description of the AEC dialog components is included below.



Activity LEDs

- **`Far`** – lights when activity is detected from the remote site.
- **`Near`** – lights when activity is detected from the local site.
- **`Update`** – lights when the AEC is updating, converging, or reconverging.

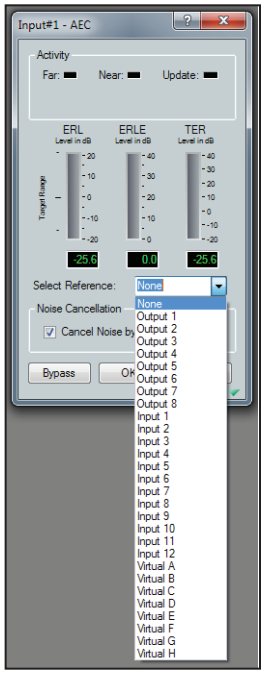
Meters

- **ERL** – the ratio in dB between the signal at the reference and the signal at the AEC channel input. When ERL is a positive number, the signal level at the AEC channel input is lower than the signal at the selected reference (0 to +15 dB is desirable).
- **ERLE** – the amount in dB of potential echo signal that the AEC algorithm, not including NLP processing, is cancelling.
- **TER** – the sum of ERL + ERLE, in dB.

Select Reference

Select the AEC reference from a drop-down list, populated with the following:

- Output channels (1 – 8)
- Input channels (1 – 12)
- Virtual Return channels (A – H)



Noise Cancellation

Noise cancellation can be switched on or off from the AEC dialog. The noise canceller detects steady state noise, such as HVAC or other continuous system noise, and effectively remove it without causing audible artifacts.

Noise cancellation is engaged or disengaged using a checkbox. When the box is checked, noise cancellation is engaged, or switched on. When cleared, noise cancellation is disengaged, or switched off. The default setting is noise cancellation switched on and set to 15 dB of noise attenuation.

Up to 20 dB of noise cancellation is available, in 0.1 dB increments.

Setting Gain Structure for AEC

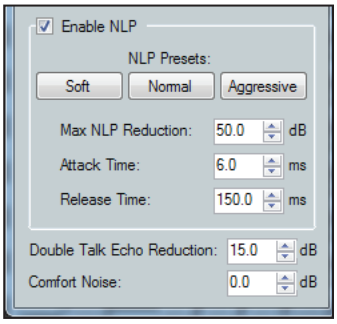
It is important to optimize the audio levels of the DMP 128 for AEC to be effective (see **Optimizing Audio Levels** on page 100).

Advanced AEC Controls

Click on the open/collapse icon at the bottom of the AEC dialog to reveal the advanced AEC controls.

Advanced control functionality is as follows:

Non-linear Processing (NLP) Controls



- `Enable NLP` — this box is selected by default. NLP is necessary for the removal of echo.
- `NLP Presets` — click a button to load a set of values to the three NLP parameters; Max NLP Reduction, Attack Time, and Release Time.
  - **`Soft`**
  - **`Normal`**
  - **`Aggressive`**

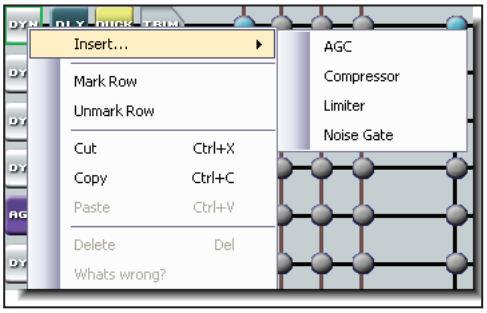
The default parameters (shown at right) match the **`Normal`** preset.

- `Max NLP Reduction` – the maximum possible reduction in echo artifacts that can be applied. The range is 0.0 to 80.0 dB in 0.1 dB increments.
  Default is 50.0 dB.
- `Attack Time` – the speed in which NLP is applied. The range is 0.0 to 100.0 msec in 0.1 msec increments.
  Default is 6.0 msec.
- `Release Time` – the speed in which NLP is released. The range is 1.0 to 3000.0 msec in 0.1 msec increments.
  Default is 150.0 msec.

Additional Controls

- `Double Talk Echo Reduction` – sets the amount of echo reduction applied during double-talk. The range is 0.0 to 20.0 dB in 0.1 dB increments.
  Default is 15.0 dB.
- `Comfort Noise` – sets a comfort noise level in dB to eliminate states of complete silence, which could be perceived as a failed connection. The range is 0.0 to 40.0 dB in 0.1 dB increments.
  Default is 0.0 dB which turns comfort noise off.

Dynamics (DYN)



A dynamics processor alters the dynamic range, the difference between the loudest to the quietest portions, of an audio signal. Each input channel provides two dynamics processor blocks that, when inserted, provide one of four types; AGC, Compressor, Limiter, or a Noise Gate processor.

Once a processor has been inserted, individual processor parameters can be changed in the dialog box, accessed by double-clicking the processor block. For comparison, the block can be bypassed by clicking **Bypass**.

All parameters are displayed in a text box and have a resolution to 0.1 increments. Parameters can be set by direct entry in the text box to replace existing text, then press <Enter>, or <Tab>, or clicking to another area. Threshold, gain/attenuation, target, and ratio parameters have adjustment points on the graph display. Use the mouse to **click**+**drag** the graph point to the desired destination or value. All time values have a horizontal slider allowing adjustment in 1 ms increments by either a **click**+**drag** of the slider handle, or focusing on the slider, then using left or right arrow keys (the <Page Up> and <Page Down> keys adjust in increments of 10 ms).

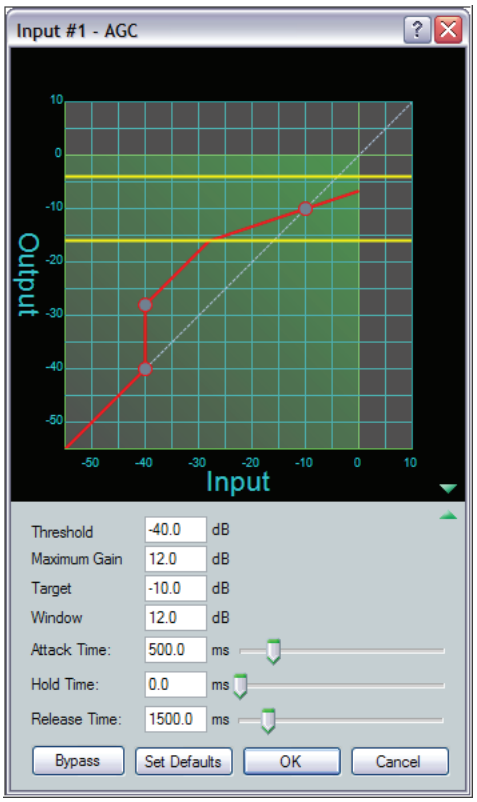
The table below lists each dynamics processor type, parameters, and factory default settings for the processor.

Parameter	AGC	Compressor	Limiter	Gate
Threshold	-40.0 dB	-30.0 dB	-10.0 dB	-65.0 dB
Max Gain	12.0 dB			
Target	-10.0 dB			
Window	12.0 dB			
Attack Time	500.0 ms	5.0 ms	2.0 ms	1.0 ms
Release Time	1500.0 ms	100.0 ms	50.0 ms	1000.0 ms
Ratio		2.0 :1		20.0 :1
Hold Time	0.0 ms	100.0 ms	50.0 ms	300.0 ms
Max. Attenuation				25.0 dB
Soft Knee		Off	Off	

Details of the individual dynamics blocks follow.

AGC (Automatic Gain Control)

AGC adjusts the gain level of a signal based on the input strength to achieve a more consistent volume. Below the set threshold, the signal is not affected. Above the threshold, weaker signals are boosted up to the maximum gain setting to reach a user-defined target level. As the signal level approaches the target level it receives less gain or no gain at all. Once the signal level reaches the target level all gain is removed. Click in each field or use the sliders to change the values.



Threshold — The input level where maximum gain is applied (after the attack time is exceeded). On the graph at right follow the red input level from the lower left to -40 dB where the first red circle is. Signal levels less than -40 dB remain at their original levels. Signal levels at or exceeding -40 dB have up to 12 dB of gain applied (**Maximum Gain**).
The threshold level can be adjusted from -80.0 to 0.0 dB in 0.1 dB increments.
Default is -40.0 dB.

Maximum Gain — The highest amplification applied to a signal exceeding the threshold and up to the lower limit of the window (see **Window** below).
Maximum Gain can be set from 0.0 dB to +60 dB in 0.1 dB increments.
Default is 12.0 dB.

Target — The desired average signal level of the output when AGC is applied. AGC can vary the gain according to the input signal level, specified target level and maximum gain. As the signal approaches the target level of –10 dB, gain is reduced until at –10 dB, gain is no longer applied.
The target level can be adjusted from -40 dB to 0.0 dB in 0.1 dB increments.
Default is –10.0 dB.

Window — Indicated by two yellow lines, is a specified range above and below the target level. Below the lower line maximum gain is always applied to the signal. When the signal reaches the window, gain control begins scaling in a linear fashion to achieve smoother results as the signal reaches the target level.

The window range can be set in 0.1 dB increments from 0.0 dB to 20.0 dB.

The default threshold is – 40 dB. The default target level is – 10.0 dB. The default window range is 12.0 dB.

Attack Time — Adjusts the time delay for AGC to engage after the input signal level reaches or exceeds the threshold level.
Attack time can be adjusted from 0.0 to 3000.0 ms in 0.1 ms increments.
Default is 500.0 ms.

Hold Time — Adjusts how long AGC continues to boost the signal after the input signal drops below the threshold and before release time begins.
Hold time can be adjusted from 0.0 to 3000.0 ms in 0.1 ms increments.
Default is 0.0 ms.

Release Time — Adjusts the time it takes to return the signal to normal (unprocessed) levels after the signal no longer exceeds the threshold level setting. Release time begins only after hold time is reached.
Release time can be adjusted from 10.0 to 10000.0 ms in 0.1 ms increments.
Default is 1500.0 ms.

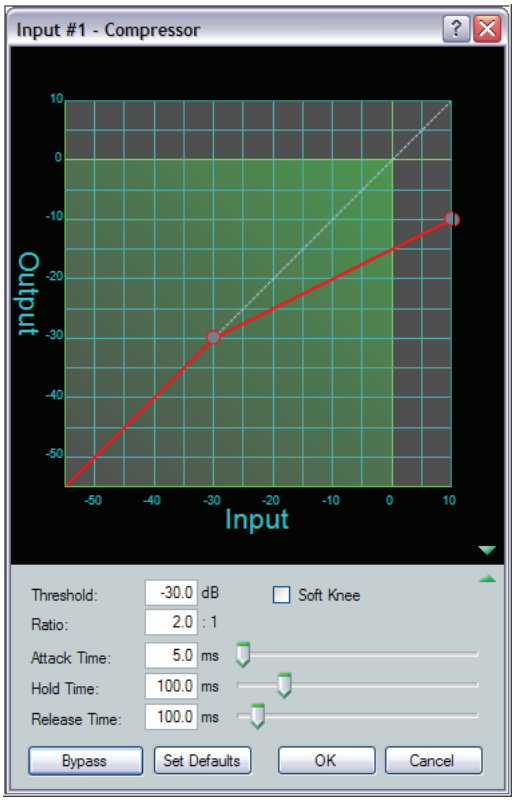
Compressor

The compressor regulates signal level by reducing, or compressing, the dynamic range of the input signal above a specified threshold. The input level to output level ratio determines the reduction in the dynamic range beyond the threshold setting. For example, with a ratio setting of 2:1, for every 2 dB of input above the threshold, the compressor outputs 1 dB.

Compression is commonly used to contain mic levels within an acceptable range for maximum vocal clarity. A compressor can also make softer sounds louder in one of two ways. The dynamic range can be reduced by compressing the signal above the threshold while raising the post-compressor gain/trim (referred to as "make-up gain"). Alternately, the input signal can be increased while the compression ratio above the threshold is increased correspondingly to prevent clipping. Both techniques have the effect of making louder portions of a signal softer while at the same time increasing softer signals to raise them further above the noise floor.

Compression can also be used to protect a system or a signal chain from overload similar to a limiter. Click in each field or use the sliders to change the values.

The default threshold is -30 dB and default ratio is 2.0:1.



Threshold — The input signal level above which compression begins (subject to attack time) and below which compression stops (subject to hold and release time).
The threshold level can be adjusted from -80.0 to 0.0 dB in 0.1 dB increments.
Default is -30.0 dB.

Ratio — The input signal level reduction when compression is engaged.
Ratio can be adjusted from 1.0 to 100.0 in 0.1 increments.
Default is 2.0:1.

Attack Time — Adjusts the time delay for compression to engage after the input signal level reaches or exceeds the threshold level.
Attack time can be adjusted from 0.0 to 200.0 ms in 0.1 ms increments.
Default is 5.0 ms.

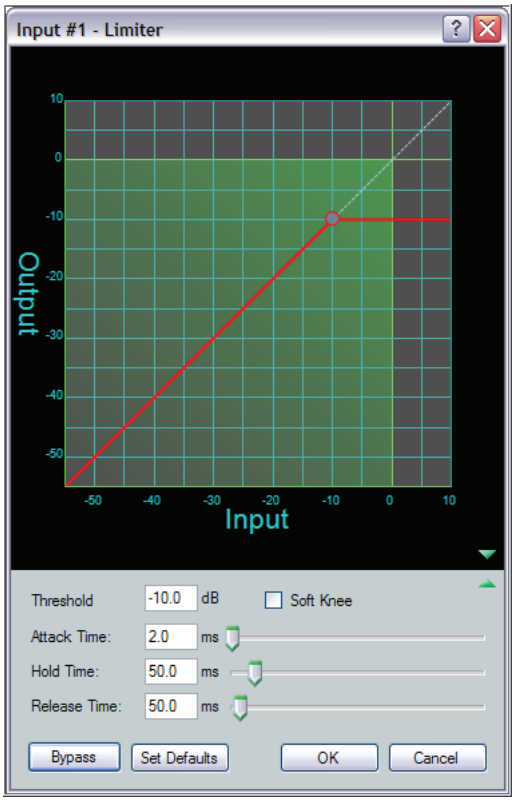
Hold Time — Adjusts how long compression continues after the input signal drops below the threshold and before release time begins.
Hold time can be adjusted from 0.0 to 500.0 ms in 0.1 ms increments.
Default is 100.0 ms.

Release Time — Adjusts the time it takes to return the signal to normal (unprocessed) levels after the signal no longer exceeds the threshold level setting. Release time begins only after hold time is reached.
Release time can be adjusted from 10 to 1000.0 ms in 0.1 ms increments.
Default is 100.0 ms.

Soft Knee — Select the **`Soft Knee`** checkbox to smooth and soften the transition from uncompressed to compressed output levels. There are no adjustments.

Limiter

The limiter restricts the input signal level by compressing its dynamic range above a specified threshold. The limiter is most commonly used to prevent clipping, protecting a system against component or speaker damage. While the limiter is closely related to the compressor, it applies a much higher compression ratio of ∞:1 above the threshold. The ratio is fixed and cannot be changed. Click in each field or use the sliders to change the values.



Threshold — The input signal level above which limiting begins (subject to attack time) and below which compression stops (subject to hold and release time).
Threshold level can be adjusted from – 80.0 to 0.0 dB in 0.1 dB increments.
Default is –10.0 dB.

Attack Time — Adjusts the time delay for limiting to engage after the input signal level reaches or exceeds the threshold level.
Attack time can be adjusted from 0.0 to 200.0 ms in 0.1 ms increments.
Default is 2.0 ms.

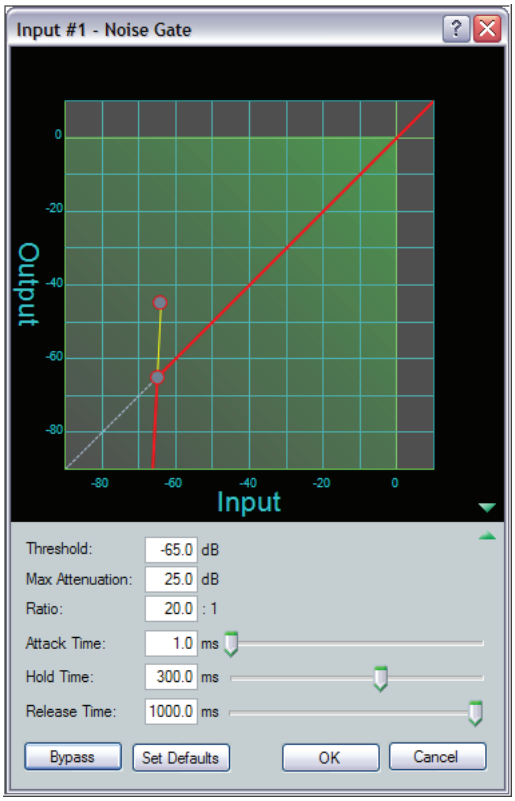
Hold Time — Adjusts how long limiting continues after the input signal drops below the threshold and before release time begins.
Hold time can be adjusted from 0.0 to 500.0 ms in 0.1 ms increments.
Default is 50.0 ms.

Release Time — Adjusts the time it takes to return the signal to normal (unprocessed) levels after the signal no longer exceeds the threshold level setting. Release time begins only after hold time is reached.
Release time can be adjusted from 10 to 1000.0 ms in 0.1 ms increments.
Default is 50.0 ms.

Soft Knee — Select the **Soft Knee** checkbox to smooth and soften the transition from uncompressed to compressed output levels. There are no adjustments.

Noise Gate

The noise gate allows an input signal to pass only when it exceeds a specified threshold level. Above the threshold level, the signal passes unprocessed; below the threshold the signal is attenuated at the rate set by the ratio adjustment. The typical setting of the noise gate threshold is just above the noise level of the environment or source equipment. That allows signals that are above the noise to pass, and attenuates the noise when there is no signal to eliminate background noise. Click in each field or use the sliders to change the values.



Threshold — is the input signal level below which attenuation (gating) begins (subject to attack time) and above which gating stops (subject to hold and release time).
The threshold level can be adjusted from -80.0 to 0.0 dB in 0.1 dB increments.
Default is -65.0 dB.

Max Attenuation — is the maximum attenuation of the signal when it drops below the threshold.
Maximum attenuation can be adjusted from 0.0 to 80.0 dB in 0.1 dB increments.
Default is 25.0 dB.

Ratio — is the input signal level reduction when gating is engaged.
The ratio can be adjusted from 1.0 to 100.0 in 0.1 increments.
Default is 20.0:1.

Attack Time — adjusts the time delay for gating to engage after the input signal level drops below the threshold level.
Attack time can be adjusted from 0.0 to 200.0 ms in 0.1 ms increments.
Default is 1.0 ms.

Hold Time — adjusts how long gating continues after the input signal drops below the threshold. If the signal is still below the threshold when hold time ends, release time begins.
Hold time can be adjusted from 0.0 to 500.0 ms in 0.1 ms increments.
Default is 300.0 ms.

Release Time — adjusts the time it takes to return the signal to normal (unprocessed) levels after the signal is no longer below the threshold level setting. Release time begins only after hold time is reached.
Release time can be adjusted from 10 to 1000.0 ms in 0.1 ms increments.
Default is 1000.0 ms.

Delay (DLY)

The delay processor block, when inserted, provides a means to delay the audio signal. Audio delay syncs audio to video or can time-align speakers that are placed at different distances from the listener. The DMP 128 can set delay by either of two criteria: time or distance (feet or meters).

The default unit setting is time with a range of 0.0 ms to 200.0 ms adjustable in 0.1 ms steps. Default is 100.0 ms.

Settings are controlled with a vertical slider and indicated in the readout field. Click within the readout field to change the value, or change the number, then press <Enter>, or press <Tab>, or click away from the field.

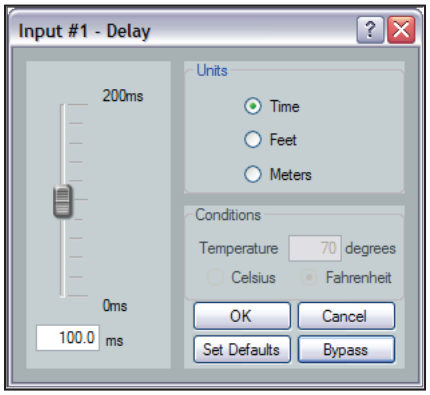


Figure 31. Delay Dialog

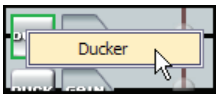
Slider adjustments made in feet or meters correspond incrementally to the distance required to make 1 ms, or 5 ms adjustments (detailed in the table below). If more precision is required, enter time in 0.1 ms increments into the readout field.

Method	Time	Feet	Meters
Click + drag	1 ms	~1.1 feet	~0.3 m
Focus + arrow	1 ms	~1.1 feet	~0.3 m
Focus + Page Up/Down	5 ms	~5.6 feet	~1.7 m

When distance (feet or meters) is chosen, the conditions (temperature) field becomes available and can be set either by degrees Fahrenheit or Celsius by clicking the appropriate selection button. When entering a distance, time delay compensation is automatically modified based on differences in the speed of sound due to air temperature. Default is 70 degrees Fahrenheit.

NOTE: When using distance (feet or meters), set a temperature value first, then set the distance.

Ducking



Ducking provides a means to duck, or lower, the level of one or more input signals when a specified source must take precedence. The ducking processor block, when inserted, provides a means to duck one or more mics and program material (ducking targets) when the processor detects a signal from the ducking source. Ducking lasts for the duration of the interrupting signal (ducking source) determined by the threshold setting (plus hold and release time) and restores the original levels of the ducked inputs once the other signal has ceased.

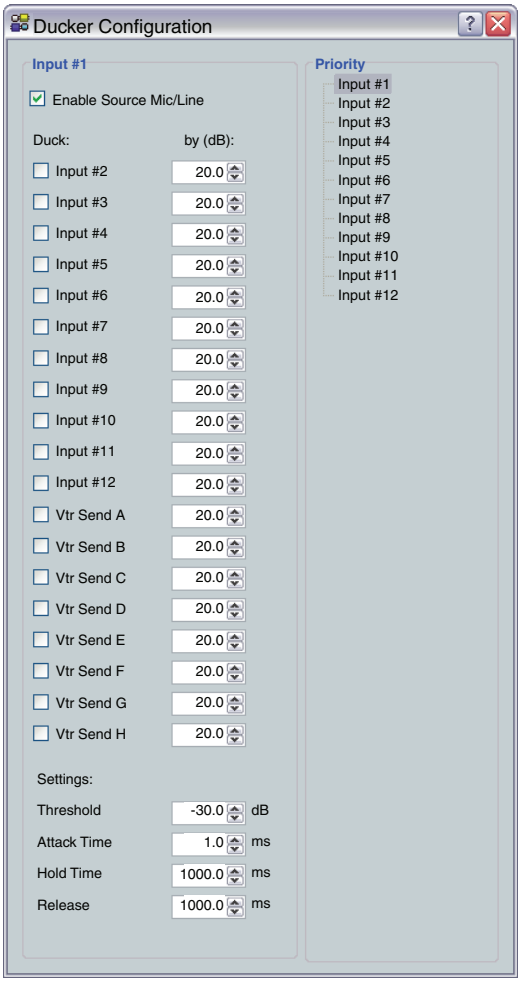
NOTE: Ducking is not functional when an input chain includes active automixing. If the input to output mix-point is orange, indicating it includes automixing, ducking will not function for that input. To enable ducking either delete the automix processor in the signal chain or uncheck **`Includes Automixing`** at the mixpoint.

Ducking is useful when:

- Program material needs to be attenuated in order to more clearly hear a narrator voice.
- One microphone, such as one used by a master of ceremonies, needs to have priority over other mics, program material, or both.
- A paging mic needs to attenuate all other signals.

All ducking processor blocks are controlled via a common dialog box that opens when any of the ducking blocks are selected. All empty ducking processor blocks have no ducking source or target settings by default.

When the first ducking source is inserted (shown at right), no ducking targets are selected.

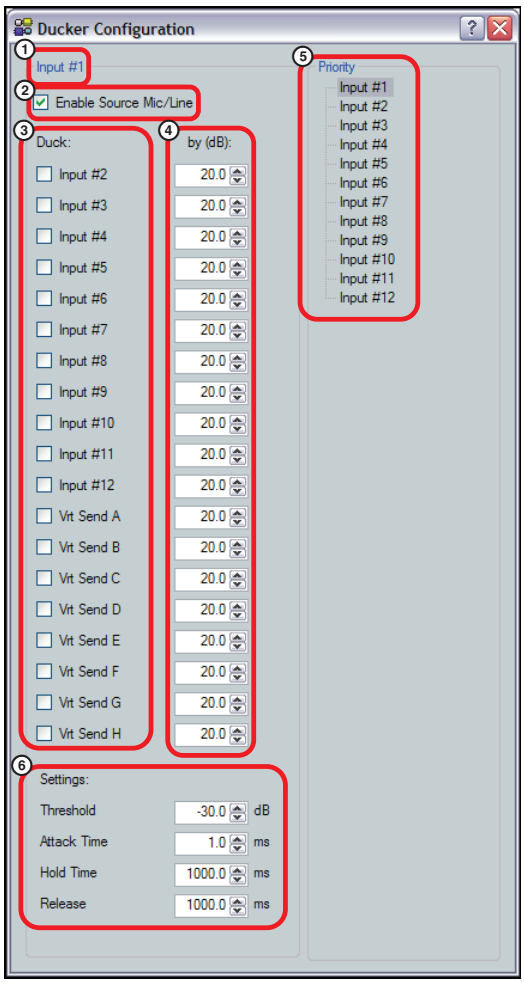


NOTES:

- Signal reduction is not cumulative. Ducking reduces an input by the amount set in the **`by (dB):`** text box next to the input selection even if it is being ducked by another ducking source (see **Ducking and Priority Ducking** on page 53).
- Duck targets do not affect signals routed to expansion outputs 1 through 8.
- Duck targets do affect signals routed to expansion outputs 9 through 16.

Ducking Configuration

Ducking is configured in a dialog box that opens when an active ducking processor block is double-clicked.



① Current Source

Shows the input selected as the ducking source. Ducker settings affect the input channel shown here. When a ducker dialog is opened, the current source defaults to that channel. The current source can also be selected via the priority readout/source selector (see below).

② Enable Source Mic/Line checkbox

When checked, ducking is enabled for the current source and the ducker processor block is lit. When cleared, ducking is disabled for the current source and the ducker processor block is unlit.

③ Duck: (targets)

Shows all potential input targets. Only inputs checked are ducked. The current source is not available as a target (a source cannot duck itself). If the current source has been designated as a target of another input channel, that input channel is not available (a target cannot be the source).

④ by (dB):

Individual attenuation settings for each duck target in dB. The default is 20.0 dB. If additional attenuation of a target is required, increase this value.
The attenuation range is 80.0 to 0.0 dB in 0.1 dB increments.

⑤ Priority

Displays the hierarchy of ducking source to duck targets. Priority levels are displayed in tree fashion. Input channels that are targets being ducked by a source are shown as indented below the source. Any input channel displayed in the tree is an active link. Click an input channel to select that channel as the current source. The current source indicator (①) reflects the selected input channel.

⑥ Settings:

Used to configure the parameter settings for the ducker source. When a ducker block is copied, these settings are transferred.

Threshold — Sets the input signal level, in dB, the ducking source must exceed before ducking begins. If ducking does not occur quickly enough to avoid loss of speech or program material from the ducking source, decrease this setting. If ducking occurs too soon, allowing background noise to trigger ducking, increase the setting.
The range is -60 to 0 dB in 1 dB increments. Default is -30 dB.

Attack Time — Adjusts the time to duck the targets once the threshold is exceeded.
The range is 0 to 3000 milliseconds in 1 millisecond increments. Default is 1 millisecond.

Hold Time — Determines the time, in milliseconds, after a ducking source signal drops below the threshold before ducking ceases.
The range is 0 to 10000 milliseconds in 1 millisecond increments. Default is 1000 milliseconds (1 second).

Release — Determines how long, in milliseconds, after the ducking source level is below the threshold and the hold time is met, the ducking targets take to restore signal levels.
The range is 10 to 10000 milliseconds in 1 millisecond increments. Default is 1000 milliseconds (1 second).

Priority

In some cases, multiple levels of ducking can be required to enable an input source to take precedence over all but one other input.

In this example, Inputs 2 through 6 are set to duck when Input #1 has a signal above the ducking threshold. Input #2 is set to duck inputs 5 and 6. Since Input #1 has previously been set to duck Input #2, Input #1 is disabled to prevent contradictory priorities.

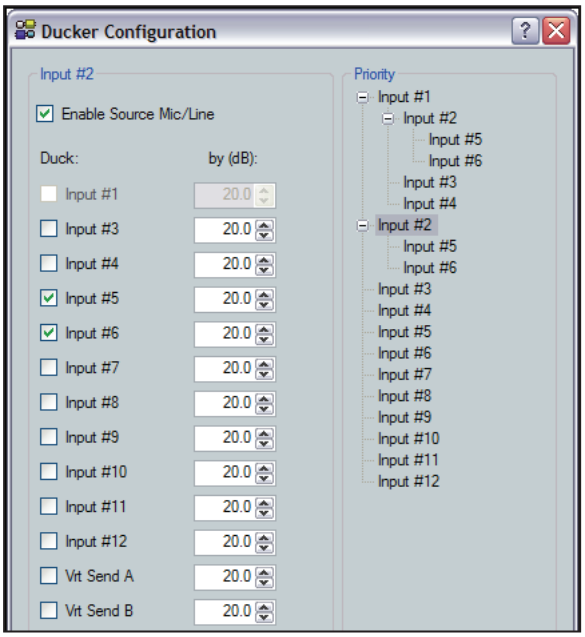


Figure 32. Ducker Configuration, Input Priority

Notice the priority tree on the right side of figure 32. The inputs are arranged by their priority status. Input #1 has all other ducked inputs under it, therefore if a signal is detected, it will trigger Inputs 2 through 6 to duck. If Input #2 detects a signal and there is no signal on Input #1, Input #2 will trigger inputs 5 and 6 to duck. However if the Input #1 signal exceeds the threshold, it will then duck all inputs including Input #2.

NOTE: Ducking attenuation is not additive. When an input target is ducked, regardless of how far down the priority line it is, the maximum attenuation is that set for the individual input and virtual send in the **"by (dB):"** column near the center of the dialog box.

See **Ducker Tutorials** on the next page for additional information.

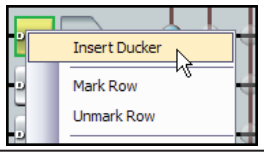
Ducker Tutorials

The examples below are based on different input configurations. Insert a ducker from a ducker processor block using one of the following methods:

Double-click the block, then click **Ducker**

-or-

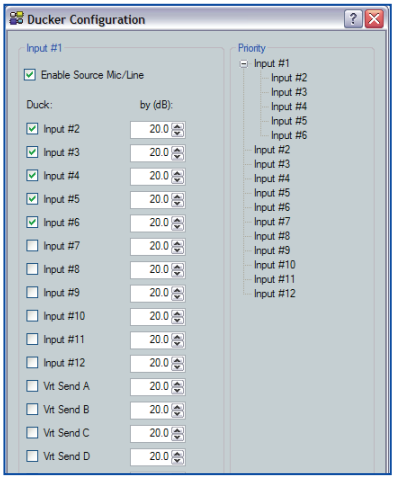
Right-click the box to open context menu, then click **Insert Ducker**



Once inserted, double-click on the ducker block to open the ducker configuration dialog box. The **Enable Mic/Line Source** box is checked.

Ducking and Priority Ducking

The first inserted mic ducks all selected targets.



To set a ducking source:

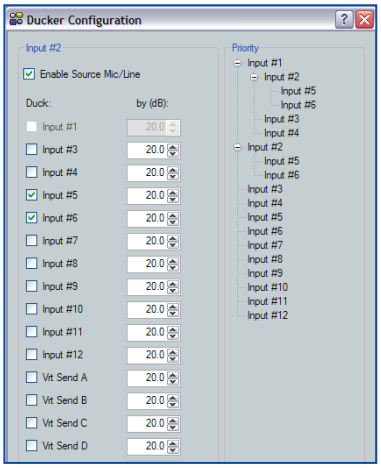
1. Insert a ducking processor on input #1.
2. Open the ducker configuration dialog box and select the desired duck targets. In this example inputs #2 through 6 are the ducking targets.
   Any signal on input #1 that exceeds the ducking threshold now ducks inputs 2 through 6.

The ducking processor also provides a means to have an additional input duck other targets using the priority feature. The second input ducks its selected duck targets, and can also be ducked by the first ducking source.

To set an additional ducking source:

1. Insert a ducking processor on the additional ducking source.

   In this example input #2 is the second ducking source, with input #1, as shown above, as the first source.



> **NOTE:** Since it was previously selected as a ducking target, Input #1 is not available as a target of input #2.

2. Open the ducking dialog box for the input and select the desired duck targets. In this example inputs #5 and #6 are the ducking targets of input #2.

   Any signal on input #2 that exceeds the ducking threshold now ducks inputs 5 through 6. The ducking targets can be changed at any time by double-clicking the input #2 ducking processor block.

   Since input #2 is a target of input #1, if a signal on input #1 exceeds the ducking threshold, inputs 2 through 6 are still ducked regardless of whether the signal on input #2 exceeds its ducking threshold.

> **NOTE:** No input will be ducked more than the amount set in the **by(dB):** box.

Automix (AM)

An automixer manages multiple microphone sources, gating or varying input gain automatically. When properly set, the automixer system will improve use and performance when multiple mics are in use. The two basic types of automixer include gated and gain-sharing.

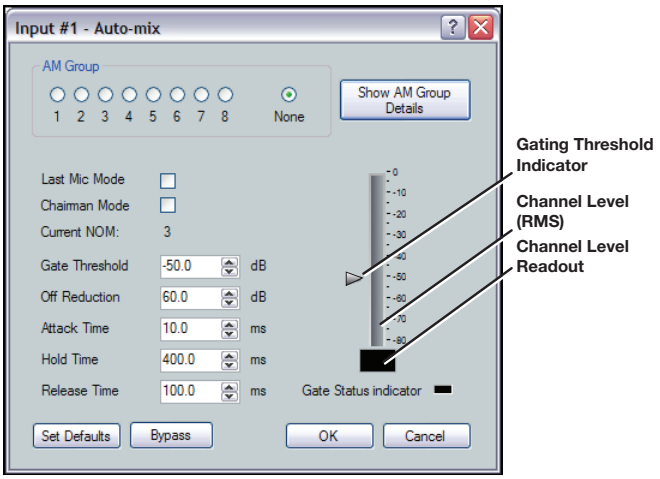
A gated automixer attenuates an input channel when the signal level drops below a user-defined threshold. DSP Configurator allows the user to divide these automixers into gating groups. Each gating group is effectively a separate automixer.

A gain sharing automixer sets a maximum room gain and splits this among all open mics, based on their input levels. While a gain sharing mixer typically has less delay in reacting to a speaker, gated automixers will normally produce a better noise floor.

The DMP 128 allows the user to choose between a gating automixer and a gain sharing automixer. When the number of open mics (NOM) is set to zero, the automixer is gain-sharing. When a NOM value is provided, the automixer is gated.

The DMP 128 uses an automix dialog box to configure the parameters of each channel and select an AM group.

Automix parameters:

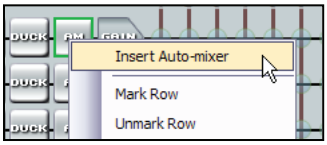


- **AM Group (Assignment)** — Assigns the channel to a gating group. Selections are 1 through 8. Default is **None**.
- **Show AM Group Details** — Accesses a dialog box that details current groups and the parameter member assignments to each (see the following section).
- **Last Mic Mode On/Off** — Prevents all mics from gating off at the same time, ensuring there is always one active mic channel. There are four possible states:
  - If not enabled on any mic input, all mics gate off.
  - If enabled on all mics, the last active mic remains on.
  - If enabled on one mic:
    - The enabled mic remains active if it is active when all other mics gate off, or
    - If the enabled mic is not active, it gates on when all other mics gate off.
  - If enabled on some but not all mics, then:
    - If enabled on the last active mic, this mic remains active, or
    - If not enabled on the last active mic, then the first enabled mic in the group gates on.
- **Chairman Mode On/Off** — One mic or multiple mics can be set to **Chairman** under the Gating Priority list. When a chairman mic is gated on, all non-chairman mics are gated off to the off reduction level.
- **Current NOM** — Displays the selection of the maximum number of mics that may be gated open at any time, per gating group. The setting can be changed using the AM Groups dialog box. Current NOM range is 1 through 12.
- **Gate Threshold** — The signal level below which the mic channel gates off and above which it gates on.
  Range is –60.0 dB to 0.0 dB.
  Default is –50.0 dB.

- **Off Reduction** — The channel attenuation when a mic channel gates off.
  Range is 0.0 dB to 100.0 dB attenuation (0 to –100 dB).
  Default: 60.0 dB.
- **Attack Time** — Sets the time at which gain is applied after a channel gates on.
  Range is 0.0 msec to 3000.0 msec in 0.1 msec increments.
  Default: 10.0 msec.
- **Hold Time** — The time that a mic remains active after the signal drops below a user-defined threshold.
  Range is 0.0 msec to 10000.0 msec in 0.1 msec increments.
  Default: 400.0 msec.
- **Release Time** — The time it takes to ramp the signal level to the **Off Reduction** value when the mic channel gates off.
  Range is 10.0 msec to 10000.0 msec in 0.1 msec increments.
  Default is 100.0 msec.
- **Gate Status Indicator and meter** — The meter provides real-time sampling of the selected AM channel with a digital readout of the current level the meter. The indicator lights when the channel is shut off.

To insert an automix block into a channel:

1. Insert an AM processing block in the desired channel.
   Either:
   - Right-click the **AM block** and select Insert **Automixer**, or
   - Double-click the **AM block** and select **Automixer**.



2. Double-click the inserted AM block to open it (see **Automix parameters:** on page 54).

   The AM block defaults to Bypass (bypass button red). Click **Bypass** to toggle the AM block to active (bypass button gray).

Automix Groups

Assigning individual automix channels to groups allows you to see and adjust all channels assigned to the group on one page. The automix group dialog provides details of all grouped and ungrouped inputs including the automix settings of each channel or mic. This provides an overview of all channels in the selected group at a glance. Individual settings can be changed without leaving the groups dialog. You can also select all **`Ungrouped items`** and see the channels currently unassigned to a group.

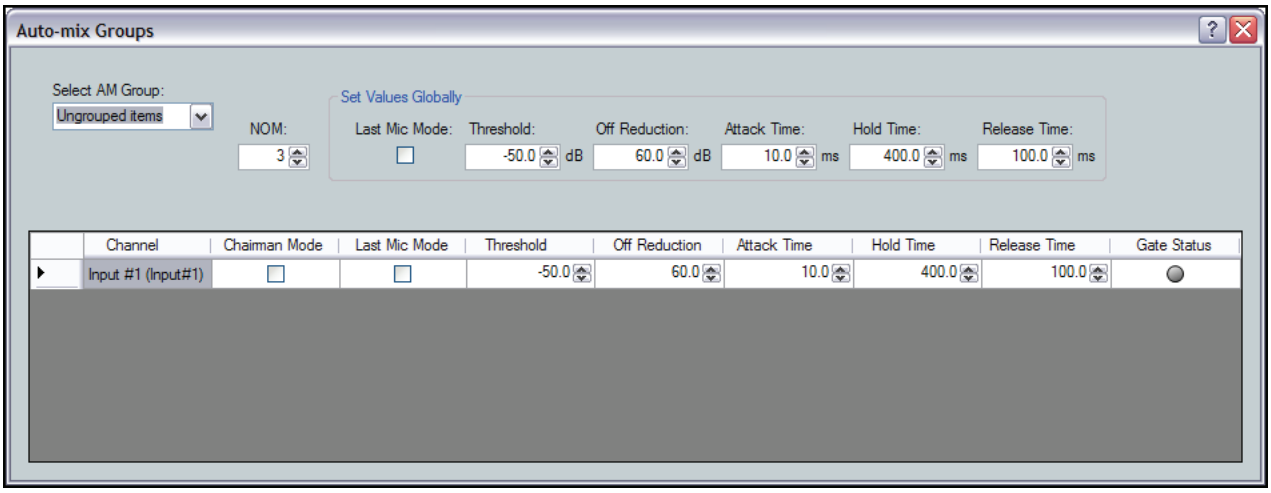


Figure 33. Automix Groups Dialog

To Configure an Automix Group:

A channel must first have an active automix block before it can be included in an automix group.

1. Insert an AM processing block in the desired channel. Either:
   - Right-click the **`AM block`** and select **`Insert Automixer`**.
   - Double-click the **`AM block`** and select **`Automixer`**.
2. Double-click the inserted **`AM block`** to open it.
3. Select the group number from a range of 1 through 8.
4. Set the parameters for the channel.
5. Repeat steps 1 through 3 for all channels in the automix group.
6. Open any AM block and click **`Select AM Group Details`** to open the automix group configuration page.
7. Set NOM (the maximum number of gated mics) for the selected group.
8. Test the system and make adjustments as needed.

   Adjustments can be made to individual automix channels using the rows by opening each individually, or globally to all channels using the AM group details page. Observe the Gate Status indicators to verify that channels gate on properly.

The AM Group Details dialog provides details of all grouped and ungrouped inputs including the automix settings of each channel or mic. This allows viewing of all channels in the selected group at a glance to provide an overview of the group. Individual settings can be changed without leaving the groups dialog.

Configuring an Automix Channel

Before configuring automix, Extron recommends that you set proper gain staging for the input mics. This ensures that adequate signal is provided for automix to work properly.

An automix block is inserted for each microphone, and the mic assigned to a group (see figure 33, Automix Groups Dialog on the previous page). Setting a reasonable NOM, the maximum number of gated mics) for the microphone group will increase intelligibility by limiting the number of open mics that are allowed to gate on at once. A NOM of three is recommended.

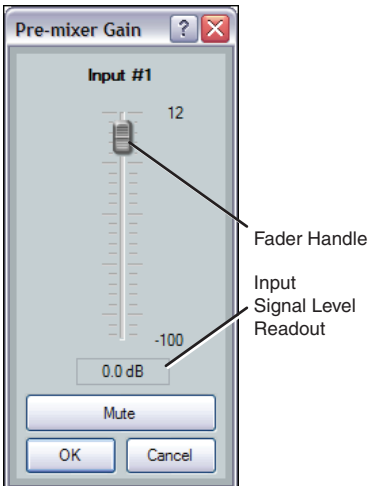
In events where a small number of talkers may need priority over other talkers (such as a presenter at a lectern) chairman mode can be enabled on the priority input channel.

Last mic mode can minimize the frequency of gate changes. This prevents rapid switching of input mics by, ensuring that a talker is not gated off when their speech is paused.

After the automixer is configured, be certain to set the appropriate mix-points to include automixing.

Pre-mixer Gain (GAIN)

The pre-mixer gain control provides gain or attenuation after the input processing signal chain. It includes a mono long-throw fader with a –100.0 to +12.0 dB gain range, and a current level setting readout below the fader. Fader adjustments are in 1 dB increments, while adjustments can be entered manually to 0.1 dB resolution.
Default is unmuted at unity (0.0 dB) gain.



Selecting the fader handle with the mouse or clicking within the fader area brings focus to the fader. The input signal level can be adjusted using any of the following methods:

- Select and hold the fader handle, then drag it to desired level in 1.0 dB steps.
- Select or tab to the fader handle, then use the up/down arrow to set the desired level in 1 dB steps.
  <Page Up> increases and <Page Down> decreases the level in 5 dB steps.
- Click in or tab to the level readout field. Type a new value, then press <Enter> or <Tab> to another area.

Telephone RX (DMP 128 C P and DMP 128 C P AT only)

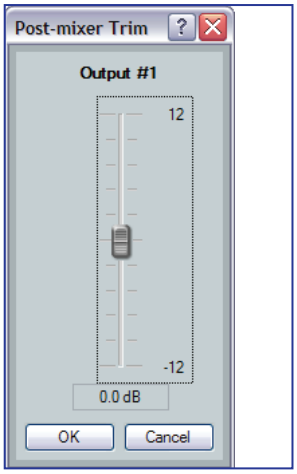
Figure 36. Telephone Rx Signal Path

The DMP 128 provides a telephone interface with separate input and output signal processing paths. The telephone input (Rx) is identical to the other input processing paths except the AEC block is not used. See **Telephone Interface** on page 126 for additional information.

NOTE: The country code must be entered before connecting the DMP 128 to a phone system.

Line Output Channels

There are eight mono line output channels. Controls and processing blocks, identical for each output channel, are described in the following sections.

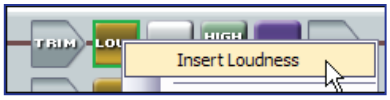


Post-mixer Trim Control (TRIM)

The post-mixer trim control provides a fader for fine adjustment of the program material prior to the output signal chain. The trim control has a range of –12 dB to +12 dB in 0.1 dB increments.

Default is unmuted at unity (0.0 dB) gain.

Loudness (LOUD)



The loudness processor block, when inserted, applies a filter compensation curve to the signal in an inverse relationship to the output volume control setting; the higher the output volume setting, the less compensation is applied (see **Calibrating Loudness** on the next page).

The loudness processor compensates for changes in human perception to varying volume levels by applying a filter compensation curve to the signal in an inverse relationship to the gain control setting. The higher the gain setting, the less loudness compensation is applied. Generally, as volume is lowered, perception of certain frequencies is progressively diminished, returning to a more flat response as volume is increased. Loudness will boost those diminished frequencies to the highest degree at low volume levels, decreasing the boost as volume increases.

Bypass must be disengaged for the loudness processor to function. The bypass button is red when engaged (loudness control defeated), and gray when disengaged (loudness control active).

When bypassed, the graph displays the current filter curve as a dotted line. When bypass is disengaged, the current filter curve is displayed as a solid line.

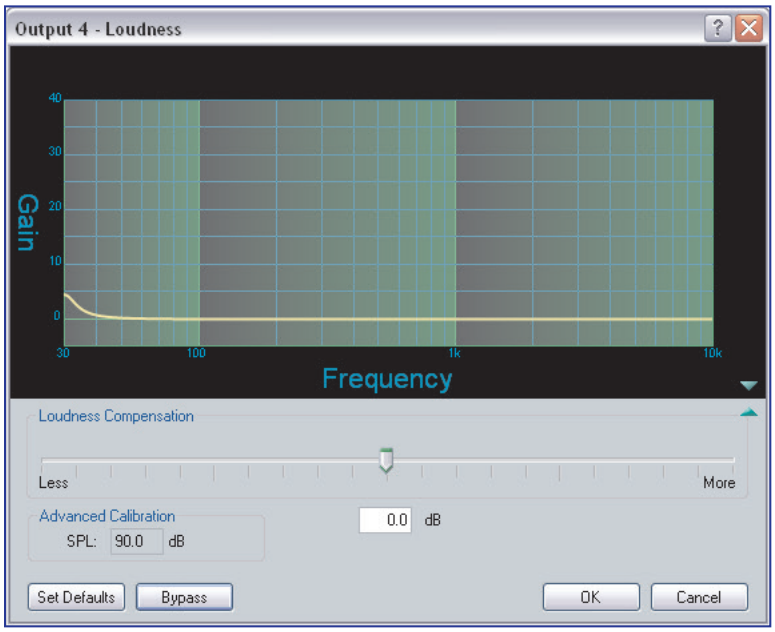


Figure 34. Loudness Dialog Dialog box

The Loudness dialog box contains the following elements:

1. **Graph** — Displays the compensation curve currently applied to the signal. These curves are read-only, and are not adjustable from the graph.
2. **Loudness Compensation slider** — From a center zero-point, the user can slide to the left for less loudness compensation (filter curve is reduced), or to the right for more (filter curve is increased). The slider position is translated into a dB value, displayed in the compensation readout box contained in the Advanced Calibration section. The slider has a 48 dB (±24 dB) range.
3. **Advanced Calibration** — Provides a value that corresponds to the position of the compensation adjustment slider. The SPL box displays the summed value of the slider and the preceding trim control.

Calibrating Loudness

The user can fine-tune the amount of loudness compensation using the compensation adjustment slider and adjusting "by ear," or by measuring SPL levels in a particular room, then using the slider to adjust the loudness filter relative to the SPL of the room and system gain structure.

Before calibrating loudness, set up the system gain structure (see **Optimizing Audio Levels** on page 100). A pre-recorded track of pink noise or pink noise from a signal generator is preferable for this purpose. Program material can also be used (using familiar material is recommended).

If using a signal generator, set it to output –10 dBu, then set the input gain of the DSP Configurator so the input meter reads –20 dBFS. If using a recorded source, the pink noise should be recorded at –20 dBFS and the player output level setting control set to maximum, or 0 dB of attenuation. For program material, set the input level to meter at approximately –15 dBFS, with peaks safely below 0 dBFS.

Unmute the mix-point from the pink noise source to the output connected to the room amplifier being calibrated. With the basic gain structure previously set up, loudness can be calibrated using an SPL meter or by ear. (Loudness can also be set using an SPL meter, then fine-tuned by ear.)

To calibrate loudness, use a sound pressure level meter set to "C" weighting:

1. Set the Loudness processor to **Bypass** (**Bypass** button red).
2. Place the meter in an average (but somewhat prominent) listening location.
3. Generate pink noise, or start the program material playback.
4. Measure the SPL in the room.
5. In the loudness dialog, adjust the slider until the value in the SPL readout box matches the reading on the SPL meter.

> **NOTE:** Theoretically, calibration can be performed with the output channel volume and post-mixer gain level set to any comfortable listening level. But a relatively loud volume (well above the ambient noise in the room) that can be easily measured is preferred.

Loudness is now calibrated. Disengage **Bypass** to hear the compensation.

Alternate method to calibrate loudness:

1. Set up the procedure using steps 1 through 3 of the previous procedure.
2. Set the compensation adjustment slider to its default center position.
3. Set the output channel volume fader to 0 dB (100% volume).
4. Adjust the amplifier until the SPL meter reads 90 dB.

Loudness is now calibrated. This method works if 90 dB is an acceptable amplifier volume limit for the room.

Setting Loudness "By Ear"

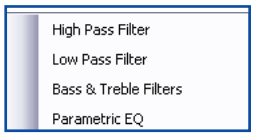
When setting loudness by ear, it is essential the system gain structure be set up first. Sit in an average (but somewhat prominent) listening location.

1. Set the loudness processor to **Bypass**.
2. Set the output volume fader in the DSP Configurator to a relatively quiet listening level. Filter compensation from the loudness processor is most prominent at low listening levels. Use familiar program material set to the levels described earlier.
3. Set the **Calibrate** slider to 0, the center point. Disengage the loudness **Bypass**. The result is a moderate enhancement to the program material, with more accentuated bass frequencies (below 500 Hz), and more brightness in the high frequencies that carry harmonic content (above 7 kHz). Engage and disengage the **Bypass** switch in order to "A/B" the difference between loudness off and on, respectively.
4. To experiment with less loudness compensation, move the loudness compensation slider to the left (less). For more loudness compensation, move the slider to the right (more).
5. Any adjustment made to the loudness compensation slider will carry through to all listening levels. Set the output volume fader in the DSP Configurator to a relatively loud listening level.
6. Engage and disengage the **Bypass** switch in order to "A/B" the difference between loudness off and on. At a loud listening level, the difference should be minimal or barely perceivable.

Delay Block (DLY)

The delay processor block, when inserted, provides a means to delay the audio signal to compensate for loudspeaker placement in situations where speakers delivering the same signal are much farther away than others. The delay processor block is identical to the delay processor available on the input (see **Delay (DLY)** on page 49). Typically the near speakers would be delayed so that audio delivery time matches the speakers further away.

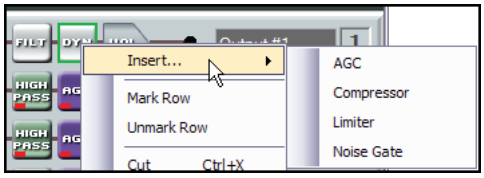
Filter Block (FILT)



The filter processor block, when first inserted, provides one of four filter selections: High Pass, Low Pass, Bass & Treble filters and Parametric EQ. Up to nine filters can be added to each filter block. The output filter block is identical to the input filter processor block except that up to nine filters total can be selected (see **Filter (FILT)** on page 30).

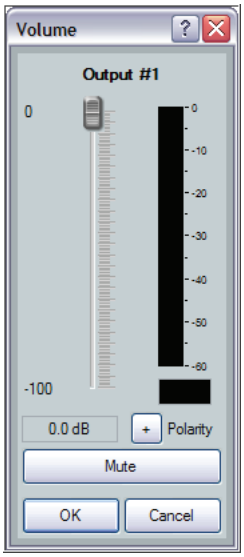
NOTE: Selecting the **Bass & Treble Filter** inserts two separate filters.

Dynamics Block (DYN)



A dynamics processor block, when inserted, provides one of four dynamics processors: AGC, Compressor, Limiter, and Noise Gate. The available processors are identical to the processors available on the input dynamics processor block and described in (see **Dynamics (DYN)** on page 44).

Volume Control (VOL)



Each output channel volume block provides a mono long-throw fader and a volume setting readout (in dB) below the fader. Volume level is adjustable with the slider or by entering the desired level directly into the volume setting readout in 0.1 dB increments.

Clicking the fader handle or clicking within the fader area brings focus to the fader. The input signal level can be adjusted using any of the following methods:

- Click and hold the fader handle, then drag it to desired level in 1.0 dB steps.
- Click or <tab> to the fader handle, then use the <up> and <down> arrow keys to change the desired level in 1 dB steps. <Page Up> (increase) and <Page Down> (decrease) the level in 5 dB steps.
- Click in or tab to the level readout field. Type a new value, then press <Enter> or <Tab> to another area.

The default setting is unmuted, at 0 dB attenuation. A peak meter displays the real-time audio level from – 60 to 0 dBFS.

Click **OK** to accept settings and close the dialog box. Click **Cancel** to ignore changes and close the dialog box.

The output volume control provides level control for each output. The output control is a trim control adjustable from –100.0 to 0 dB. The default setting is unity gain (0.0 dB).

The **Polarity** button, allows the polarity of the wires connected to the audio connectors (+/tip and -/ring) to be flipped in order to easily correct for miswired connectors.

The **Mute** button silences the post-meter audio output. When the audio output is muted, the mute button lights red, and red indicators in the block turn on.

If the output has been grouped with other inputs or outputs, the group number is indicated on the right side of this button (see **Line Output Channels** on page 58).

Telephone Tx (DMP 128 C P and DMP 128 C P AT only)

Figure 35. Telephone Tx Signal Path

The DMP 128 provides a telephone interface with separate input and output signal processing paths. The telephone output (Tx) is identical to the other output processing paths. All signals routed to the Telephone Tx are also available on output 8.

See **Telephone Interface** on page 126 for additional information.

NOTE: The country code must be entered before connecting the DMP 128 to a phone system (see **Telephone Configuration** on page 126).

Virtual Bus Returns

There are eight mono virtual bus return inputs, fed by the virtual sends. Channel controls and processing blocks described in the sub-sections that follow are identical for each virtual bus return channel.

The eight returns are divided into two similar paths. Channels A through D contain a feedback suppression processing block in each channel. Channels E through H are identical except there are no feedback processing blocks.

Virtual Bus Returns, A-D

The virtual bus is used when additional processing of an input signal is required. It is also useful to apply identical filtering, dynamics processing, loudness compensation, or signal gain/attenuation to multiple inputs.

Feedback Suppressor (FBS)

The Feedback Suppressor (FBS) is used when there is indication of feedback during live operation. Dynamic filters automatically detect feedback on a live mic channel, and engage a set of up to 5 fixed and 15 dynamic filters to counteract the frequency peaks at the detected feedback frequency. Up to 15 separate filters can be employed at any time. The 15 filters act in a FIFO, or first in, first out rotation. If all 15 filters are employed, when an additional feedback frequency is detected it overwrites the first detected feedback frequency and so on.

To avoid a new feedback frequency overwriting a previously detected one, up to five of the dynamic feedback frequencies can be placed into fixed filters. Once written into the fixed filters, the feedback frequency can only be overwritten by manually writing a new frequency to the filter.

The FBS dialog box has three tabs; **Settings**, **Dynamic Filters** and **Fixed Filters**. Global settings and view options are controlled from the **Settings** tab. Dynamic to fixed filter allocations are handled from the **Dynamic Filters** tab. Filter parameters can be modified from the **Fixed Filters** tab.

The FBS dialog box provides the following global buttons:

- **`Clear All`** — Clears all dynamic filter settings.
- **`Lock`** — Locks the dynamic filters to current settings, preventing automatic updates. This temporary mode is useful while testing the system, or during the time when dynamic filters are being converted to fixed filters. When the FBS dialog box is closed, lock mode is automatically disengaged.
- **`Bypass FBS`** — Turns off feedback detection when engaged (button is red). Only the dynamic filters are bypassed. Fixed filters remain active.
- **`Set Defaults`** — Click once to return the FBS to default settings.

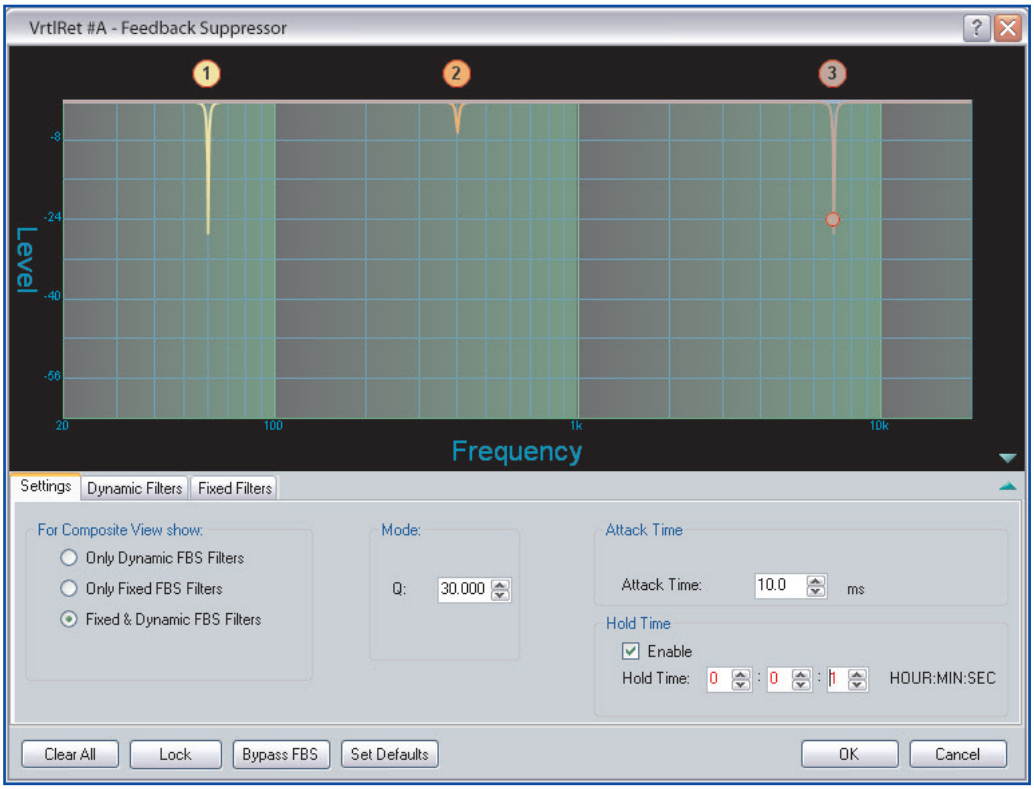


Figure 37. Feedback Suppressor

FBS Settings Tab

The **Settings** tab enables selection of the feedback suppressor parameters.

- **For Composite View show:** — The graph view is set by one of three radio buttons:
  - **Only Dynamic FBS Filters**
  - **Only Fixed FBS Filters**
  - **Dynamic & Fixed FBS Filters** (default)
- **Mode: Q** — Adjusts the notch filter Q used by dynamic filters. Similar to Q on the parametric equalizers, Q changes the bandwidth of the filter. The default setting can be modified in **Tools > Options**. The range is from 5 to 65. Larger values provide less change to the audio frequency response while lower values may provide greater feedback suppression but with more possible impact to the tonal response of the source audio.
  Suggested values for specific applications are:

Q Value	Application
7	Voice with considerable feedback potential
30	Voice with less feedback potential
65	Music with minimal feedback potential

- **Attack Time** — Sets the time at which dynamic filters are generated after feedback detection. A longer attack time (greater than 200 ms) reduces the chance that music or audio content will trigger the dynamic filters to respond. A shorter attack time (less than 2 ms) reduces the time between when feedback is detected and suppressed.
- **Hold Time** — Expressed in hours:minutes:seconds up to 9 hours. Hold time sets the time a dynamic filter setting persists before the filter is cleared. When hold time is disabled (checkbox cleared) dynamic filters persist indefinitely unless cleared manually or the device is power cycled.

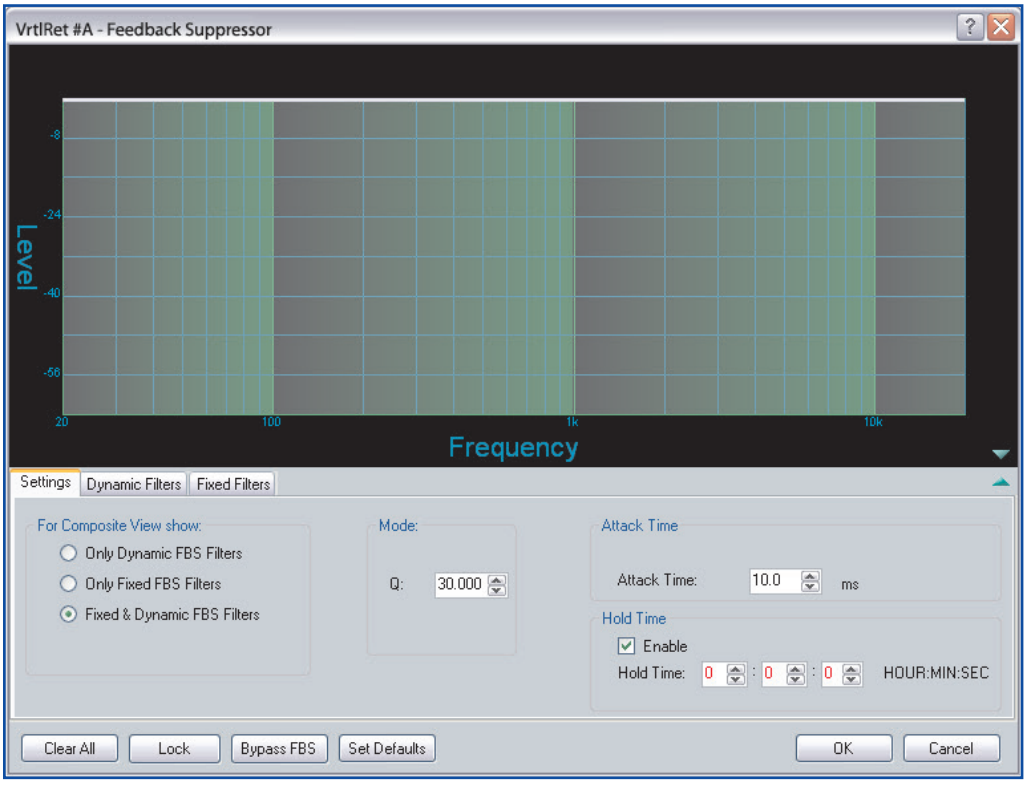


Figure 38. FBS Settings Tab

FBS Dynamic Filters Tab

This dialog contains the fifteen dynamic filters, with a scroll bar to display filters hidden due to the dialog box size.

Dynamic filters are cut only notch filters, providing attenuation up to 30 dB at the specified Q. The default Q is set in the **Tools > Options** menu, but can be changed on the settings tab prior to engaging the FBS dynamic filters.

NOTE: Changing the Q setting in the options menu after dynamic filters have been generated clears all dynamic filters.

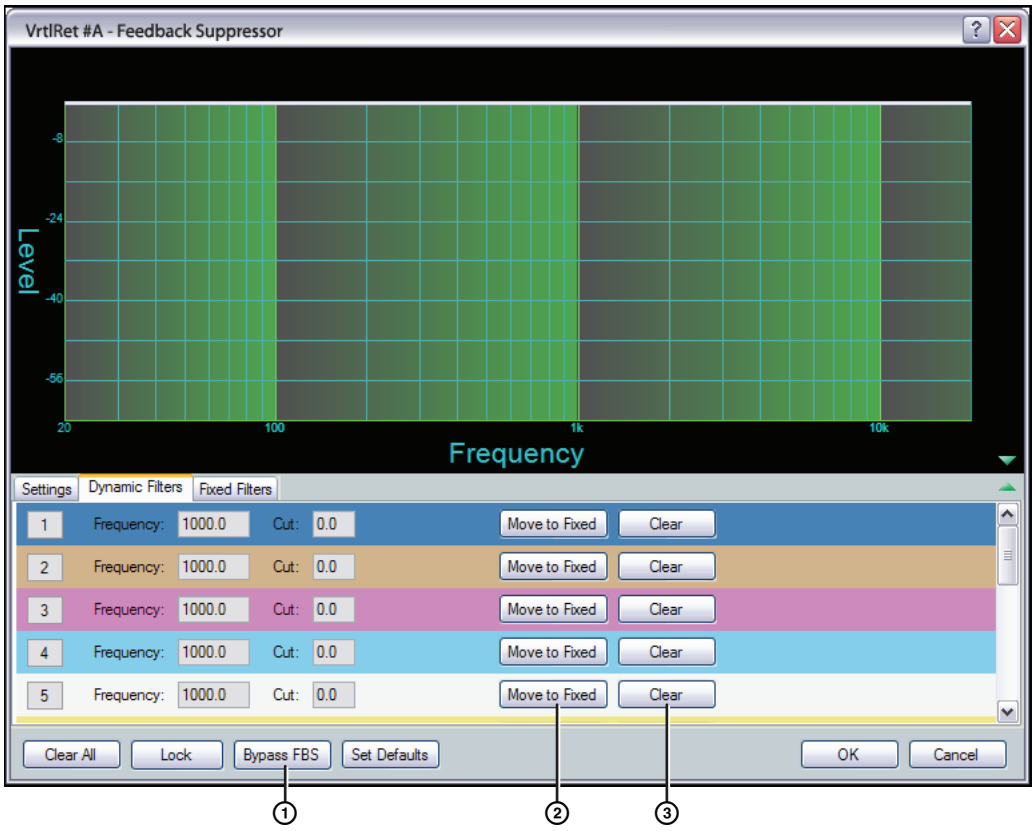


Figure 39. FBS Dynamic Filters Tab

Frequency and cut values are read only. Dynamic filters are in auto-detect mode when the FBS block is active (see figure 39, ①; **Bypass FBS** off). If testing reaches a point where no further changes are desired, the lock button can be engaged. The lock mode of operation is temporary, and is intended to be used during setup of the FBS. When the FBS dialog box is closed, lock mode is automatically disengaged.

If there are specific dynamic filters that you want to assure are not overwritten, press the **Move to Fixed** button (see figure 39, ②) to write the designated filter settings to the first available filter in the Fixed Filter tab.

NOTE: When a dynamic filter setting is moved to the fixed filter, it automatically clears that frequency from the dynamic filter.

The **Clear** button (see figure 39, ③) removes a detected frequency from the corresponding dynamic filter. A cleared filter reverts to auto-detect mode unless **Lock** mode is engaged.

FBS Fixed Filters Tab

Fixed filters are notch filters with an adjustable center frequency and Q, and up to 30 dB of cut. The fixed filters are typically set by converting dynamic filters to fixed, however adjustments to filter parameters can be manually made from the Fixed Filters tab.

Fixed Filters are inactive and the filter type is set to **Unused** by default (see rows 4 and 5 in figure 40).

Figure 40. FBS Fixed Filters Tab

No filter parameters are displayed when the filter type is set to **Unused**. As a filter is moved to the fixed filter tab from a dynamic filter, the filter becomes active and displays **Notch** as the filter type. The parameters copied from the dynamic filter are displayed in the same line. Once a fixed filter is active, settings can be modified or adjusted as needed. Fixed filters can also be individually bypassed by clicking **Bypass**.

FBS Settings Ranges and Fixed Filter Defaults

FBS Parameter	Settings Range	Default Setting
Frequency	20 Hz to 20 kHz	N/A
Q	5.000 to 65.000	30.000
Attack Time	0.0 ms to 1000.0 ms	10.0 ms
Filter Hold Time	0 seconds to 9 hours	00:00:00; Disabled

Fixed Filter Parameter	Settings Range	Default Setting
Frequency	20 Hz to 20 kHz	1000.0 Hz
Q	1.000 to 65.000	30.000
Cut	Up to 30 dB cut	0.0 dB

Filter (FILT)

Filter function and interface is identical to the mic/line input channel Filter block except that only three filters are provided (see **Filter (FILT)** on page 30).

Dynamics (DYN)

There is one dynamics processor block available on each virtual path. Dynamics function and interface is identical to the mic/line input channel Dynamics block, (see **Dynamics (DYN)** on page 44).

Loudness (LOUD)

There is one loudness processor available on each virtual path. The loudness function and interface is identical to the Output channel Loudness block (see **Loudness (LOUD)** on page 58).

Bypass must be disengaged for the loudness processor to function. The bypass button is red when engaged (loudness control defeated), and gray when disengaged (loudness control active).

Delay (DLY)

Audio Delay syncs audio to video or to time-align speakers that are placed at different distances from the listener. The Delay function and interface is identical to the input channel delay block (see **Delay (DLY)** on page 49).

Gain (GAIN)

Each virtual input channel gain block provides a mono long-throw fader with a –100.0 to +12.0 dB gain range, and a level setting readout below the fader. Fader behavior is identical to the Pre-mix-point gain block, described in the mic/line input section (see **Pre-mixer Gain (GAIN)** on page 57). Fader adjustments are in 1 dB increments, while adjustments can be entered manually to 0.1 dB resolution.
Default is unmuted at unity (0.0 dB) gain.

Virtual Bus Returns, E-H

There are four additional mono virtual bus return inputs. Virtual Bus Returns E through H are identical to A through D except there are no feedback processors.

As with the virtual bus returns A through D, these returns are used when additional processing of an input signal is required. It is also useful to apply identical filtering, dynamics processing, loudness compensation, or signal gain/attenuation to multiple inputs.

Output Mix Matrix

The DSP architecture contains an output mix matrix that connects all inputs to the line outputs, a virtual send mix matrix that connects all inputs to the virtual outputs, and an expansion (EXP) output mix matrix that connects the mic/line inputs and virtual bus returns to the expansion outputs (see figure 41 on the next page).

The output mix matrix sets mix levels from the post processing inputs and post processing virtual returns, to each line output bus. The mix-point behavior is shown on page 73.

Each mic/line input and virtual bus return is connected to a mix-point for each of the eight line outputs. In general, mix levels are set relative to each other, achieving a desired blend of input signals at an optimal output level, close to, but not exceeding 0 dBFS at the line output volume block level meter (while accounting for processing that may occur in the line output signal chain).

NOTE: Although the virtual Output and return lines, A through H, are shown as end points on the DSP Configurator screen, they are connected A to A, B to B, C to C, D to D, E to E, F to F, G to G and H to H. These connections cannot be changed.

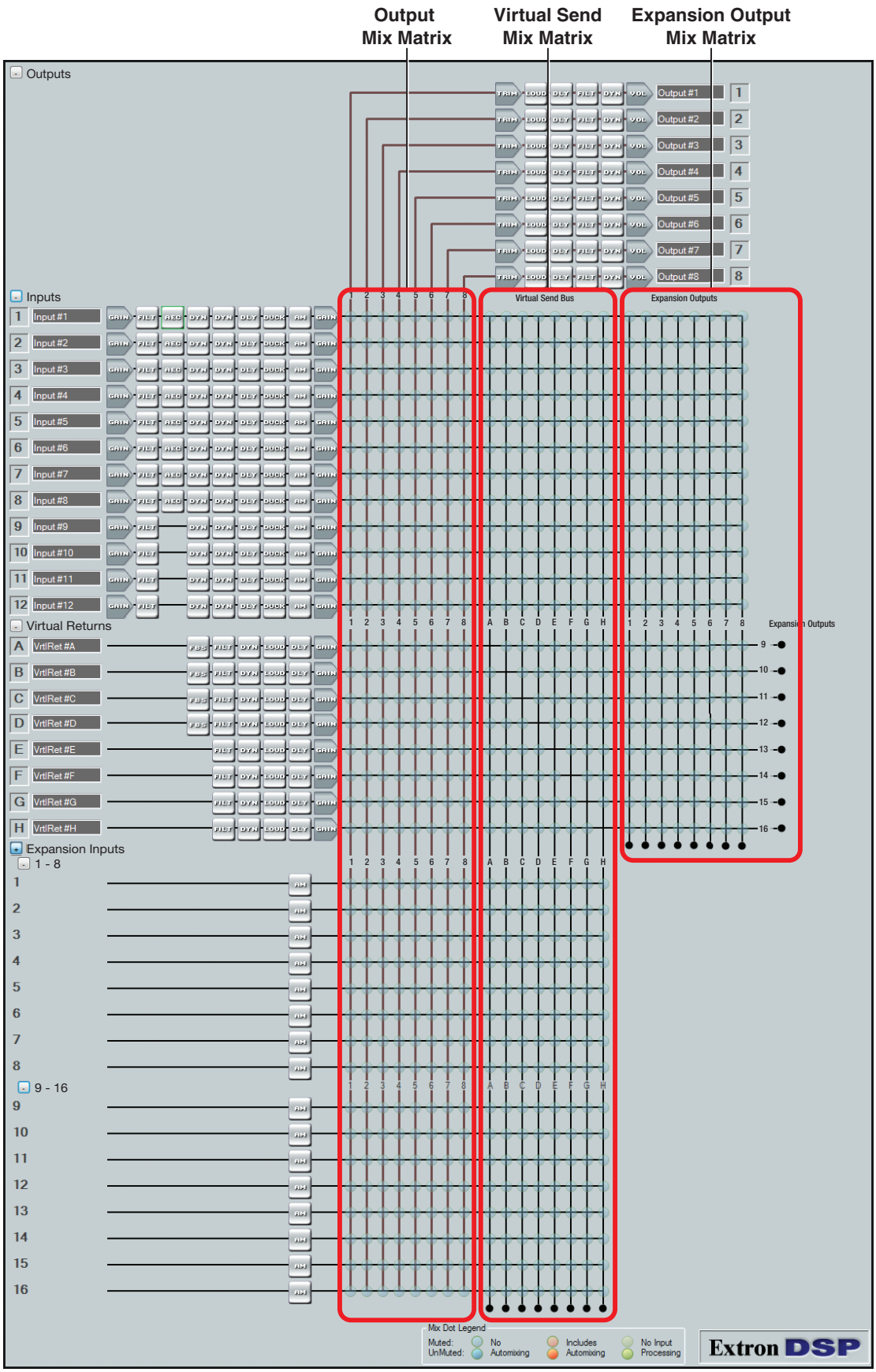
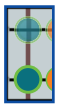


Figure 41. Overview of DSP 128 Mix-matrix

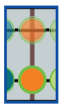
Mix-point Behavior:

Mix-point color — There are three colors of mix-points:

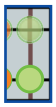
Teal indicates standard processing (default).



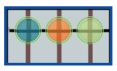
Orange indicates that the signal chain includes an auto-mix processor.



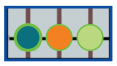
Green indicates that all signal processing has been bypassed, post input gain control.



No mix information — A faint transparent circle (teal, green, or orange) on the mix-point indicates that it is muted (contains no mix information).



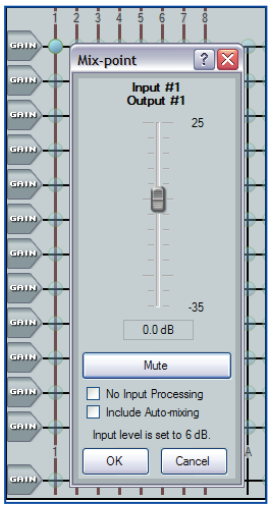
Mix information — A solid circle indicates that the mix-point contains mix information (the mix-point is unmuted).



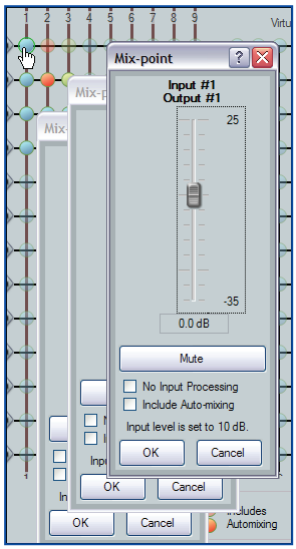
Mouse-over — The cursor changes to a hand when a mouse-over occurs at a mix-point whether the mix-point contains mix information or not.

Single-click — A single-click (click) brings focus to (selects) the mix-point, indicated by a dark green outline around the circle.

Double-click — A double-click opens the mix-point dialog box. The focus outline turns light green to indicate the open dialog box. If the mix-point is muted, the mix-point circle is gray and the **Mute** button in the dialog box is red.
If unmuted, the bubble is teal and the mute button in the dialog box is normal (typically gray for most color schemes).

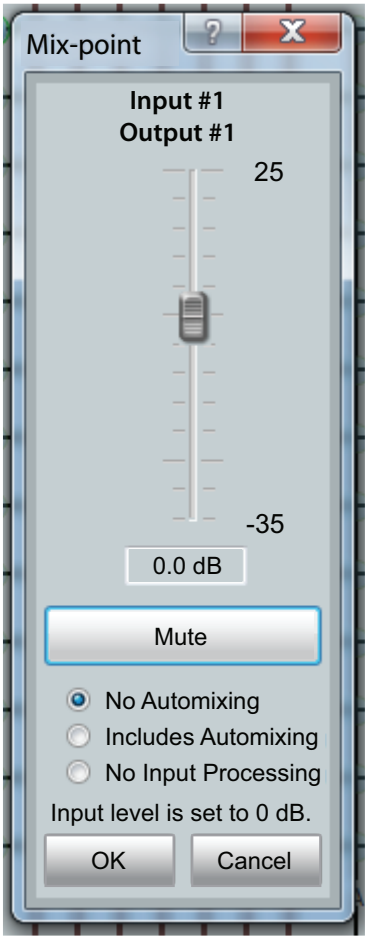


Multiple open dialog boxes — When multiple mix-point dialog boxes are open, the mix-point for the most recently opened dialog box receives the light green focus circle, while previously opened dialog boxes relinquish their focus. Focus can be returned by a click on a previously opened dialog box, or by double-clicking a mix-point.



Click a mix-point to bring focus to that mix-point. A circle appears around the teal mix-point, which remains transparent. Double-click a mix-point to open a configuration dialog box with the following components:

- **Mono Fader** — Sets the signal level from the selected input to the output bus. Gain range is -35 dB to +25 dB. Fader behavior is identical to the input channel gain block described in the mic/line input section with the exception that course adjustment (<Page Up> and <Page Down>) increases or decreases in 5 dB increments.
- **Mute** — Mutes and unmutes the signal to the output bus. The mix-point ball is transparent when muted (Mute button red) and solid when unmuted.
- **No Automixing** — When selected, bypasses all automix channel inputs. The mix-point is teal.
- **Includes Automixing** — When selected, includes the automix channel input. The mix-point turns solid orange when the mix-point is unmuted and transparent orange when muted.
- **No Input Processing** — When checked, bypasses all processing for the preceding input string. This allows a direct comparison of sound between the unprocessed signal and fully processed. The mix-point turns green when unmuted and transparent green when muted. Default is cleared.
- **OK/Cancel** — click **OK** to accept changes and close the box. **Cancel** ignores changes and closes the dialog box.



The title above the fader reflects the input and output channel names for the mix-point. The example on the right is the **Input #1** to **Output #1** mix-point set to 0.0 dB.

The input level text below the mute button indicates the input level setting for the input gain control or mute status of the selected input signal path, in this example 0 db.

Only when the mix-point is unmuted does the circle become solid.

NOTE: The **No Input Processing** and **Includes Auto-mixing** buttons are mutually exclusive. You cannot select both. If you are including an automix channel in the signal path, when you select **No Input Processing**, **Include Auto-mixing** clears and does not turn back on even when **No Input Processing** is unselected. If you want to continue to have an automixing channel in the signal path, it must be selected again.

Mix-point Examples

In order to better understand how mix-points work, the following diagrams provide examples of mixes.

NOTE: To simplify the diagrams not all input and output lines are shown.

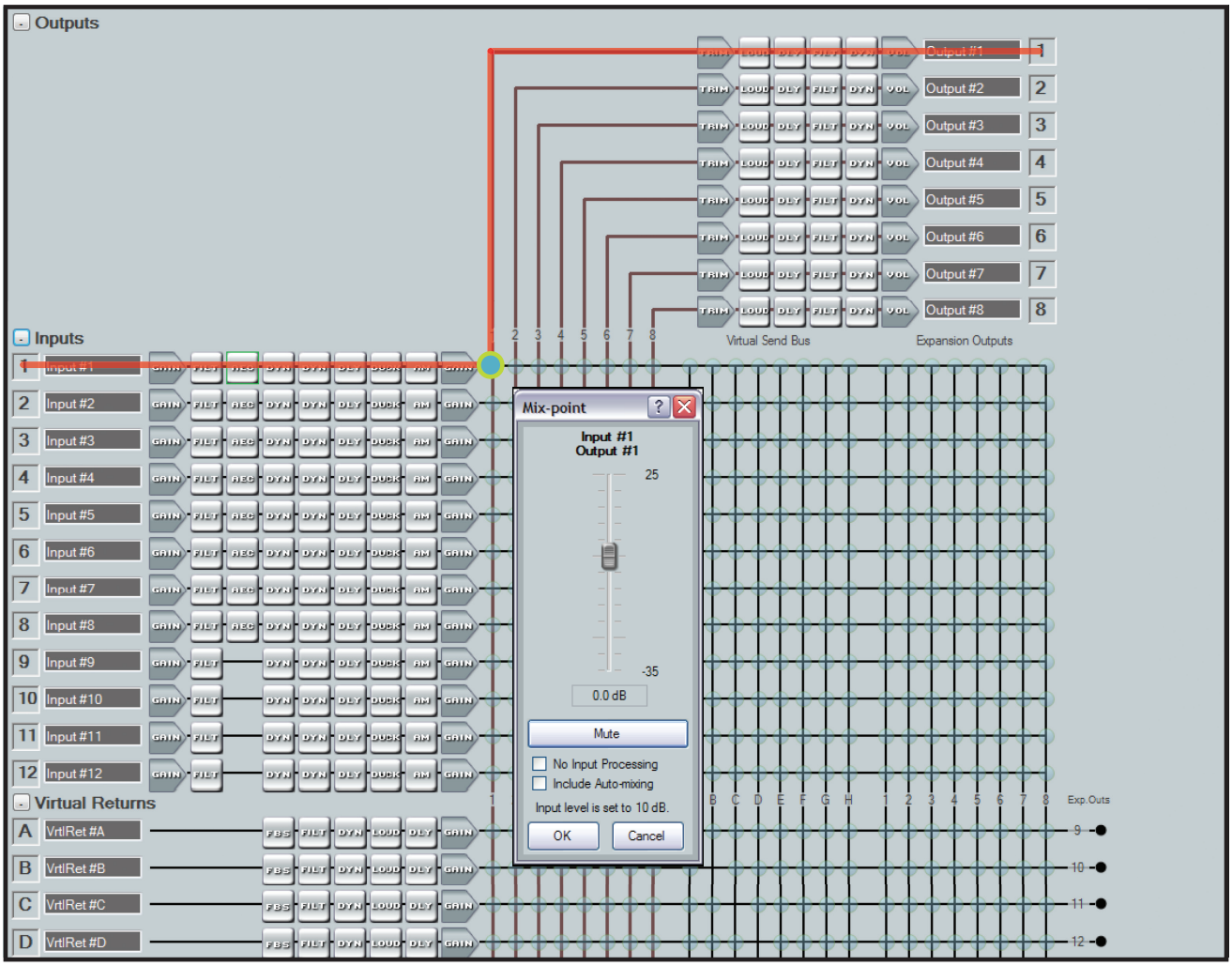


Figure 42. Input 1 to Output 1

In the first example (see figure 42) input audio from Input 1 is processed and arrives at the output 1 matrix mix-point. Double-click the mix-point to open the dialog box. When the mix-point is unmuted, the mix junction turns teal with a light green circle to indicate the open mix-point dialog box is the focus, and the signal is routed to output 1.

The mix level can be adjusted using the slider or by direct input of a value between –35.0 and 25.0 dB into the dialog box below the slider.

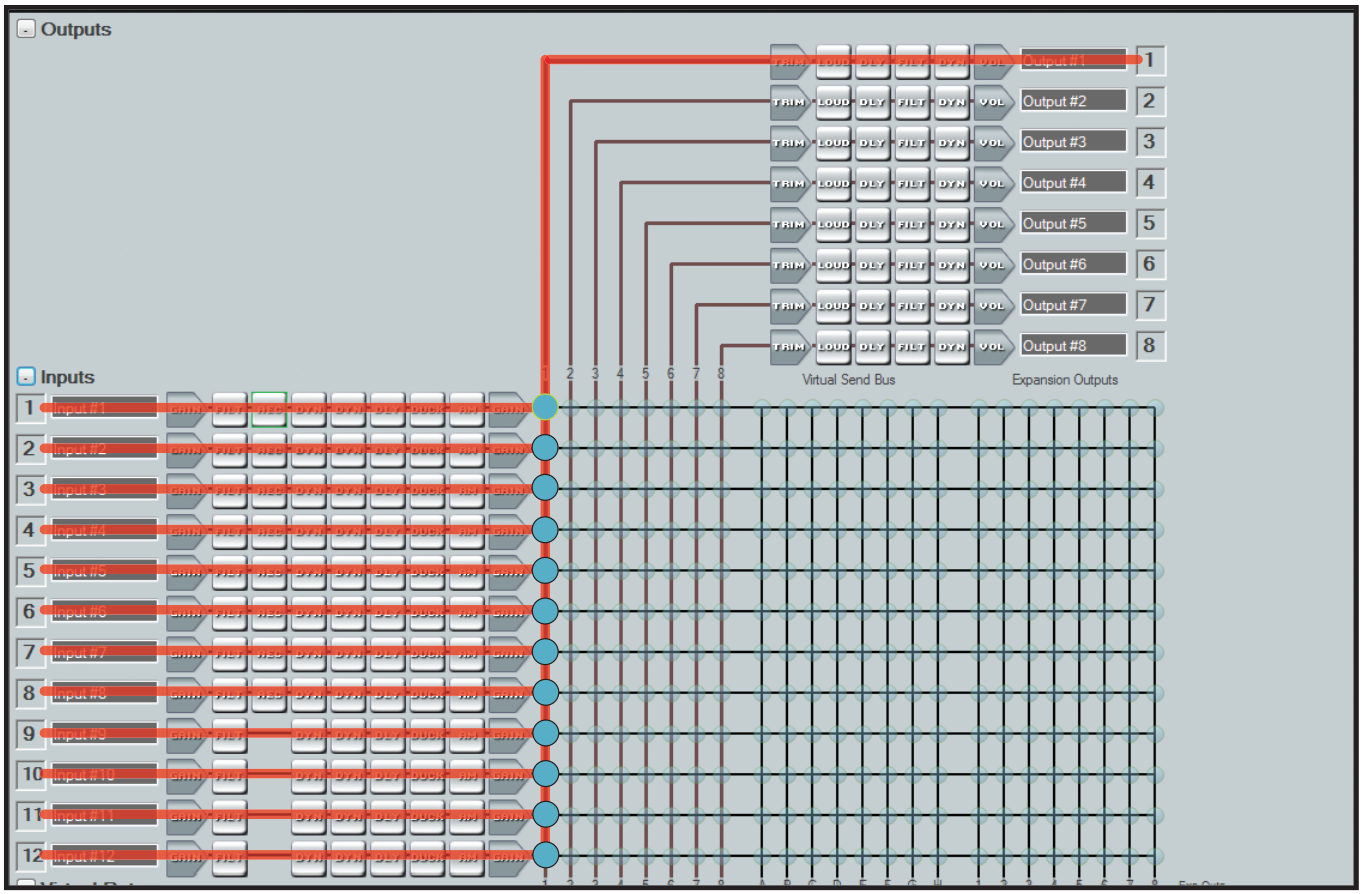


Figure 43. All Inputs to Output 1

In the next example (see figure 43), input audio from all twelve mic/line inputs are processed individually and arrive at their output 1 mix-points. As each mix-point mute button is released, its output 1 mix-point junction turns teal, and the signals are all routed to Output 1. Since all the signals are now on output signal line 1, open the individual mix-points to adjust signal levels for the desired balance. Open the output 1 trim, processing, or volume to change the signal levels or effects for the signals coming from the mix-points.

In this manner, any single input, or any number of inputs can be routed to any single output or any number of outputs.

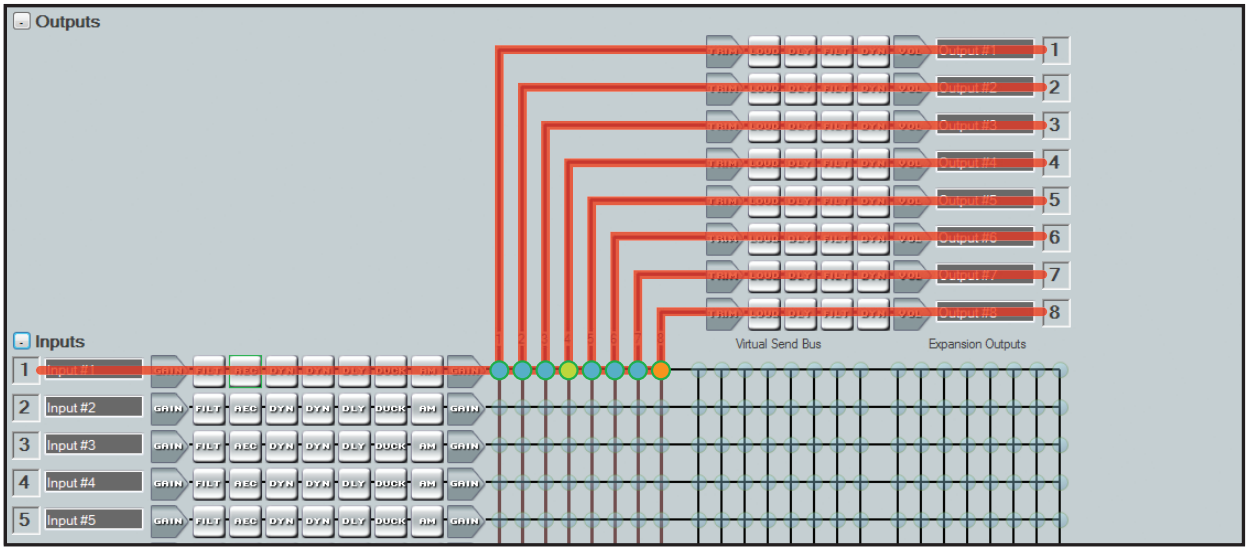


Figure 44. Input 1 to All Outputs

In the example in figure 44, input 1 has been routed to all eight outputs by unmuting the mix-point for Input 1 for each output (1 through 8) bus. The example also shows the mix-point for output four with input processing bypassed (green) and the mix-point for output eight with active automix.

Virtual Send Bus Mix Matrix

The DSP architecture contains a Virtual Send Bus mix matrix that connects the inputs and virtual bus return signals to the virtual sends. There is an additional mix matrix to route EXP input signals to the virtual sends.

The DSP Configurator main screen provides control of the virtual bus mix matrix, used to set levels from input signals to the virtual sends. Each of the twelve mic/line and eight virtual return inputs are connected to a mix-point for virtual bus A through H (and the EXP inputs). Each mix-point is muted and set to 0.0 dB (unity gain) by default. In general, mix levels are set relative to each other, achieving a desired blend of input signals at an optimal level close to, but not exceeding 0 dBFS at the output volume level meter.

The virtual mix matrix contains a section (see figure 45 below) that allow the virtual bus returns to be routed back to the virtual bus matrix for further processing using an additional virtual bus processing block. To prevent feedback loops, a virtual channel is prevented from being routed back to itself by eliminating the mix-point that would allow that to occur.

In situations requiring extra processing, the virtual bus return output is routed back to the virtual bus send mix matrix, which then routes the signal back to a processing signal chain other than the one it was routed from.

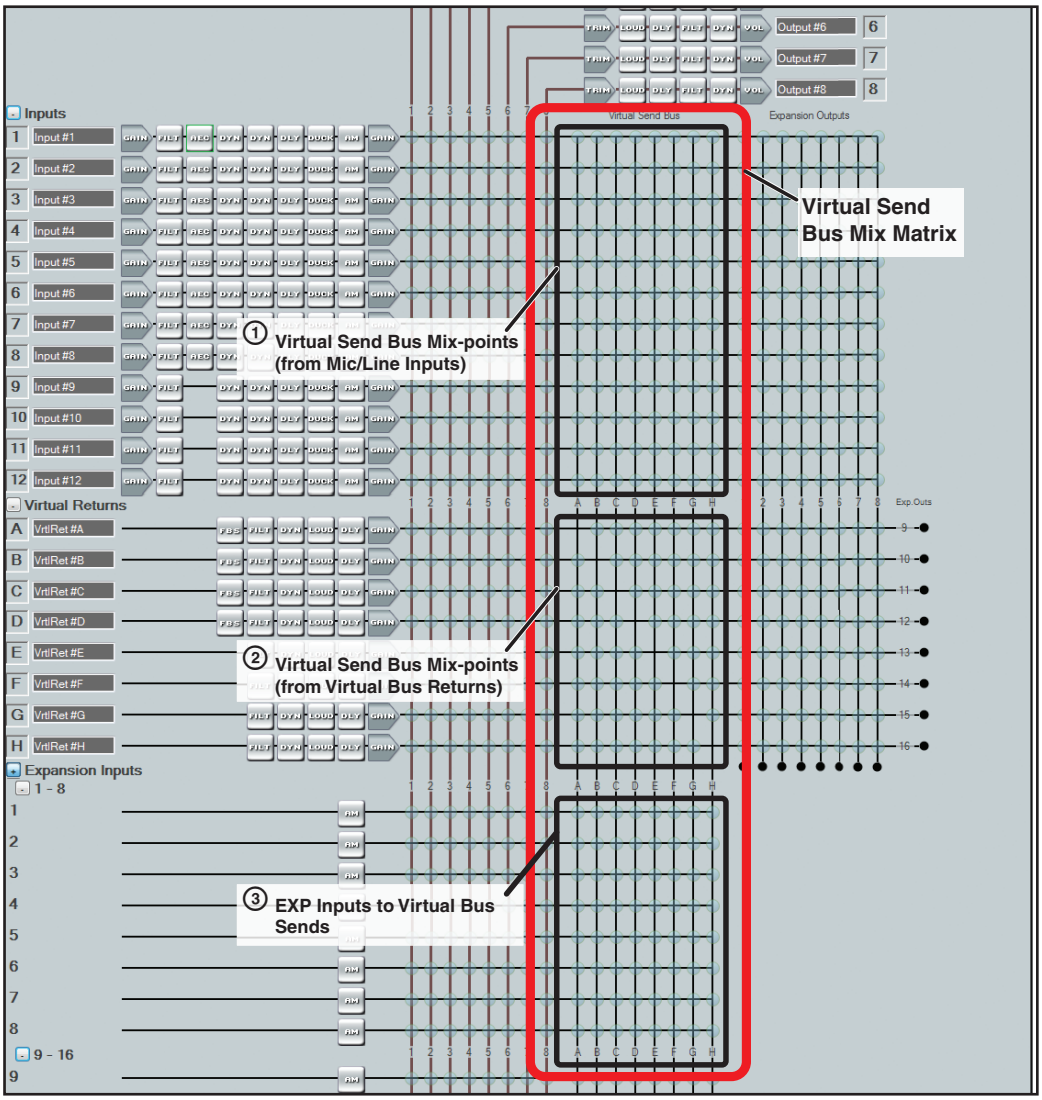


Figure 45. Virtual Bus Mix Matrix (EXP inputs 9-16 not shown)

In the example in figure 46 below, input 1 is sent to the virtual send bus by muting all eight signals on the Input 1 output mix-points and unmuting virtual send bus output 1. The virtual bus now serves as additional signal processing for the input. The signal routes from virtual send A through the virtual bus A signal chain before it is sent to the virtual bus return mix-point and finally to output 1.

This configuration is useful when more than one input requires identical processing. For example if all inputs were normalized but required a uniform gain to bring them up to adequate output levels, rather than changing each pre-mix gain control by a similar amount, all twelve inputs can be routed to a virtual bus (in this case virtual bus A). Then, using the virtual bus A return gain control, a single adjustment can apply the same gain to all twelve inputs before sending the signal to the desired output.

In other cases, when multiple mic inputs are mixed with program material, only the program material might require loudness contouring. So the mics are routed directly to the output but the program material input is routed to the virtual bus return where loudness contouring is applied. The program material is then routed to the same output as the mics.

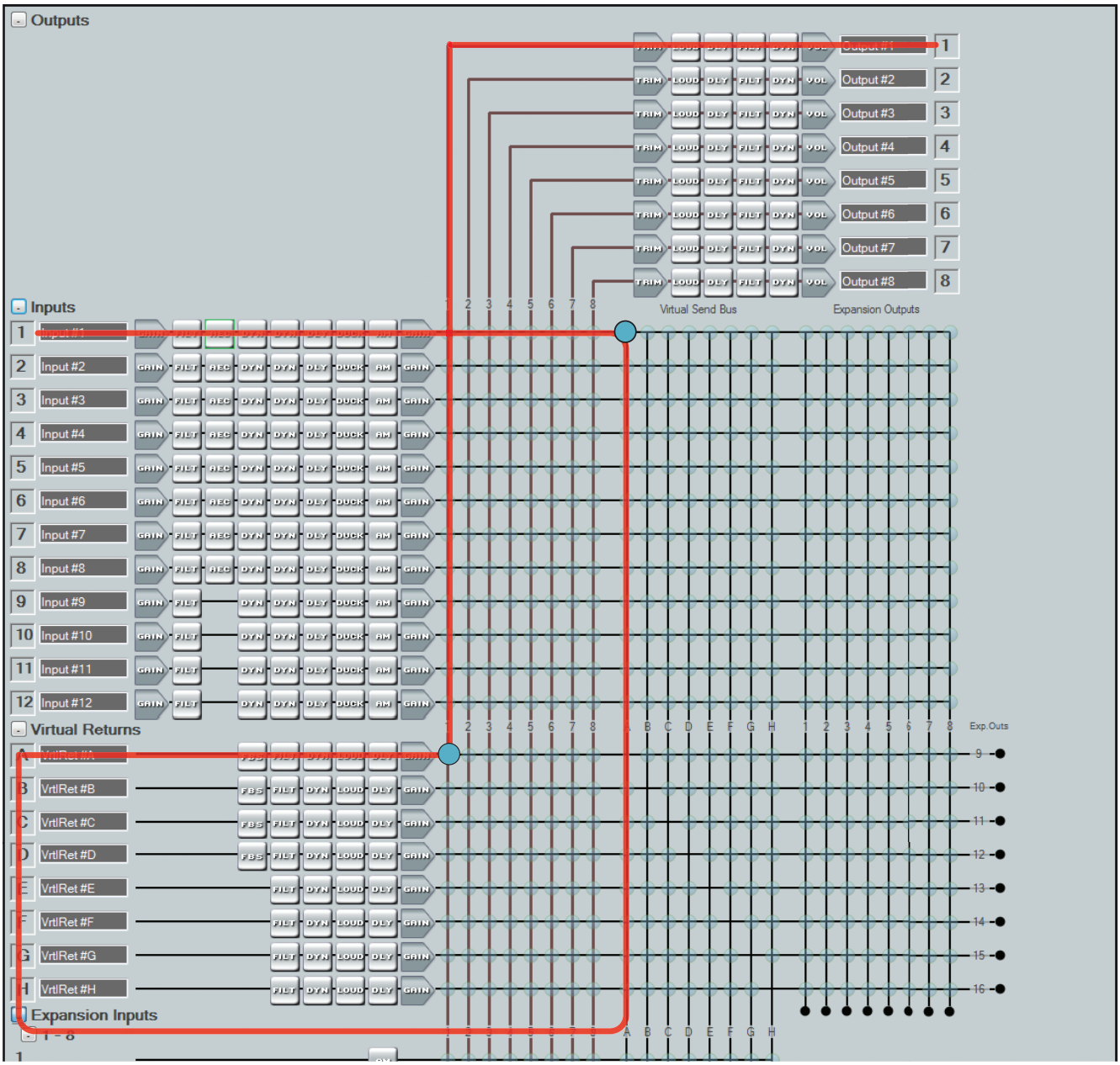


Figure 46. Input 1 to Virtual Bus A

Expansion Outputs Mix Matrix

The DSP architecture contains a third mix matrix that supports connection and control of a second DMP 128 using the included shielded CAT 6 cable. The output connects the mic/line inputs and virtual returns to the Expansion Outputs. The DSP Configurator main screen provides all necessary control of the mix matrix.

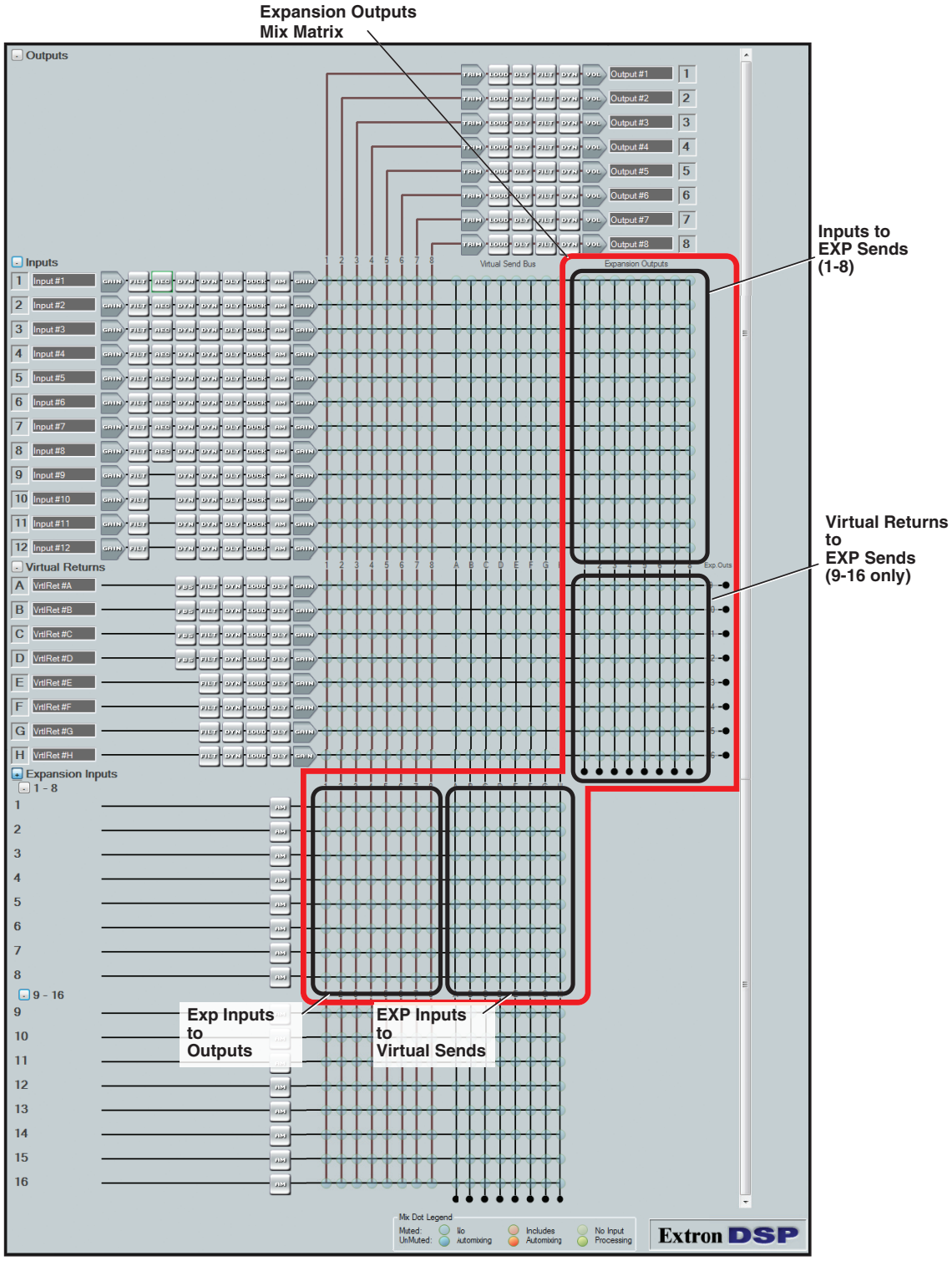


Figure 47. Expansion Bus Mix Matrix

Extron EXP Bus

Using the Extron Expansion port (EXP), two DMP 128 units can be connected together for bi-directional communication of 16 channels of audio.

The expansion bus mix matrix can route any or all of the mic/line inputs to any or all of the expansion outs (1 through 8). In addtion, there are eight channels (expansion outs 9 through 16) directly connected to the virtual bus returns.

Connecting the EXP Ports

Minimal setup is required for EXP port communications. One DMP 128 must be set as the Primary Unit, and the other as the Secondary Unit to synchronize the sampling clocks of the two units. This has no bearing on how audio is transported from unit to unit, making it unimportant which unit is set to primary.

NOTE: You **can** connect a DMP 128 AT to a non-AT unit by the EXP ports, however be aware that the AT unit defaults to the primary unit and cannot be changed.

You **cannot** connect a DMP 128 AT model to another AT model by the EXP ports because there is no option to set one as primary and the other as secondary as required.

To enable the expansion bus between two units:

The EXP bus requires one device be configured as the primary unit and the other as the secondary unit. The tools menu is used to configure the connection.

1. Power on both units. Open DSP Configurator and connect **`Live`** to the first unit. From the main menu, select **`Tools>Expansion Bus>Primary Unit`**.

   A checkmark appears beside the active unit.

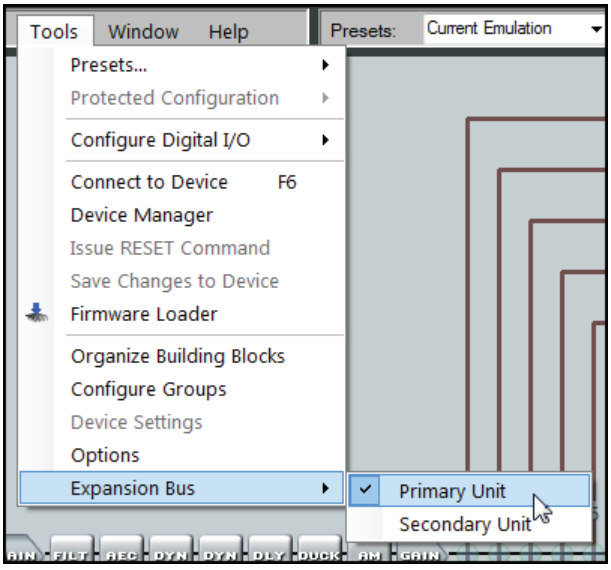
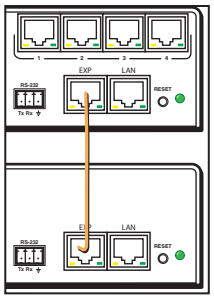


Figure 48. Expansion Bus Unit Selection

2. Select **`Tools>Connect to Device`** and connect **`Live`** to the second unit.
3. Select **`Tools>Expansion Bus>Secondary Unit`**.

   A checkmark appears beside `Secondary Unit`.

4. Connect the EXP port of one unit to the EXP port of a second unit using the included shielded (or similar) CAT 6 cable.



NOTE: The front panel EXP LED indicates device to device connection and configuration status as follows:

- **(non-AT models)**
  - **Off** — The unit is not connected to a second DMP 128.
  - **On** — The unit is connected to another DMP 128 and configured as the primary unit.
  - **Blinking** — The unit is connected to another DMP 128 and is currently configured as the secondary unit.
- **(AT Models)**
  - **Off** — Dante device is not responding.
  - **On** — The EXP port is connected to a non-AT DMP 128 and configured as the primary unit.
  - **Blinking** — The EXP port is not connected.

Using the Expansion Bus

After configuration and connection, the two units have 16x16 channels of bi-directional audio communication.

- The expansion bus from the primary unit (see figure 49, ①) sends audio to Expansion inputs 1–16 of the secondary unit (see figure 49, ②).
- At the same time, the expansion bus from the secondary unit sends audio to Expansion inputs 1–16 of the primary unit.

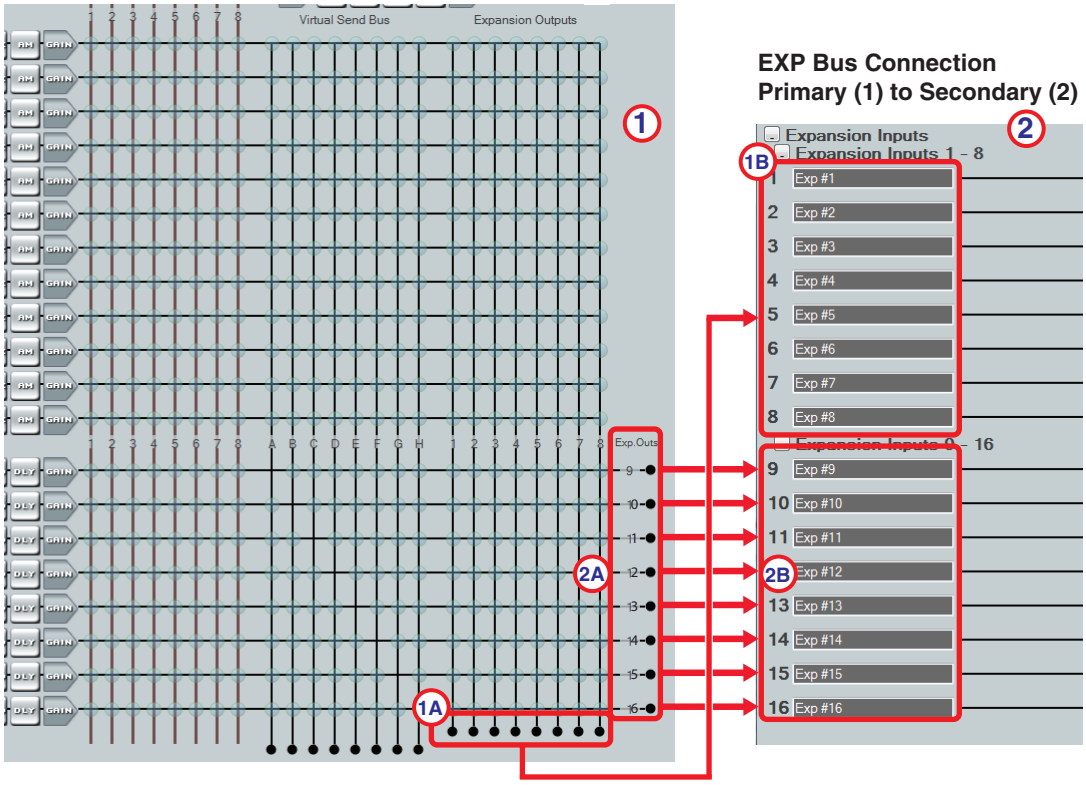


Figure 49. EXP Bus Connections

The mic/line inputs and the virtual bus returns make up the expansion bus mix matrix that feed EXP outputs 1 through 8 (see figure 49, ⓵ⓐ) of the primary unit. They are connected to EXP inputs 1 through 8 of the secondary unit (see figure 49, ⓵ⓑ), respectively.

The primary unit EXP outputs 9–16 (see figure 49, ⓶ⓐ) are direct feeds from virtual bus returns A – H (post processing). They are connected to EXP inputs 9 through 16 of the secondary unit (see figure 49, ⓶ⓑ)

Device Manager

When multiple DMP 128s are connected, use the Device Manager to select the device open in the DSP Configurator dialog box.

From the menu bar, select **`Tools > Device Manager`** to open the device manager dialog shown in figure 50.

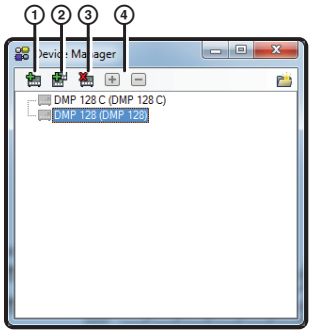


Figure 50. Device Manager Dialog

The icons function as follows:

① **`Add Device`** – Brings up the opening DSP Configurator connection dialog allowing the selection of the model number of the secondary device.

② **`Clone Device`** – Provides the option to duplicate the primary device configuration to the connected secondary device.

③ **`Delete Device`** – Deletes the highlighted device

④ **`Expand or Collapse Device`** – If the device connection tree is collapsed, allows it to be expanded. If the device tree is collapsed, expands it.

The icons for the devices are unavailable if the device is offline.

AT (Dante) Bus

For connection and software operation of the AT bus and Dante Controller software, see **Dante Controller Software Installation** on page 113.

Group Masters

There are 32 Group Masters that can each be configured to simultaneously control up to 16 group members. Group masters are configured in DSP Configurator and saved in the device. Working in emulate mode, group masters are saved in a configuration file and pushed to the device upon connection.

A group master can either be a gain control or a mute control. Only one control type can be selected as group members for control by a group master. For example, a group master can be configured to control post-matrix gain levels, but not post-matrix gains plus input gain block. A group member can, however, be controlled by multiple group masters. It is recommended this feature be used cautiously, as “overlapping” membership can quickly become unmanageable.

Group master gain controls can send specific values, such as those sent by a fader control. Group master gain can also be set by increment and decrement (see **Tools** on page 86).

Group Members

Once a group has been created, the group members — the individual controls that comprise the group — update to indicate they are now part of a group. Group members can be controlled individually, allowing for relative levels between members to be fine-tuned. Group member levels can also be set by a preset recall.

Grouped Controls

Grouping is convenient when multiple controls require muting at the same time or when multiple signal levels need to be increased or decreased simultaneously. For example, in a system with several audio outputs dedicated to a single room, the operator can want all outputs to change at the same rate and at the same time. The output 1 through 4 volume controls are grouped into a master so that one group volume control, controls the volume throughout the room.

For further flexibility, individual volume controls in the group are set for an output level based on its use. When the group fader moves, all four output control faders move in tandem while retaining their levels relative to each other.

Grouped faders move together at relative levels to the top or bottom of their travel (see figure 51, next page). If one fader reaches the limit of its travel first, it retains that position while the other faders continue to travel. When the grouped faders travel in the reverse direction, the fader that was at its limit reverts to its position relative to the other faders.

NOTE: If a block was previously muted when the group mute is activated, that block remains muted when the group mute is released.

TIP: When including a control in multiple groups, do so with care. Overlapping group membership can quickly become unmanageable. Use presets to set individual faders to known levels.

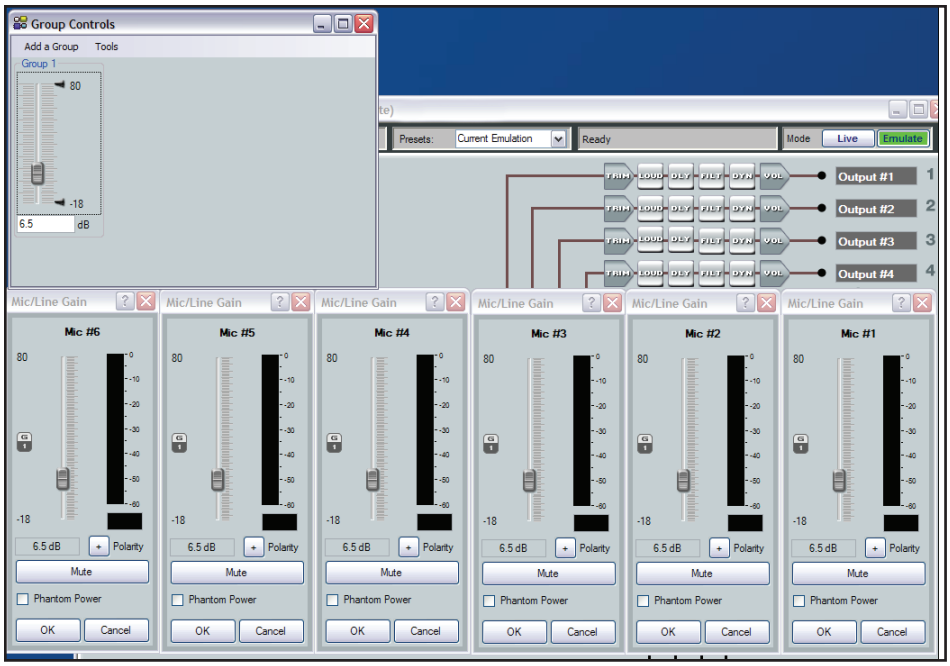


Figure 51. Sample Gain Group Master and Associated Gain Controls

Mute controls within the blocks can also be grouped (see figure 52).

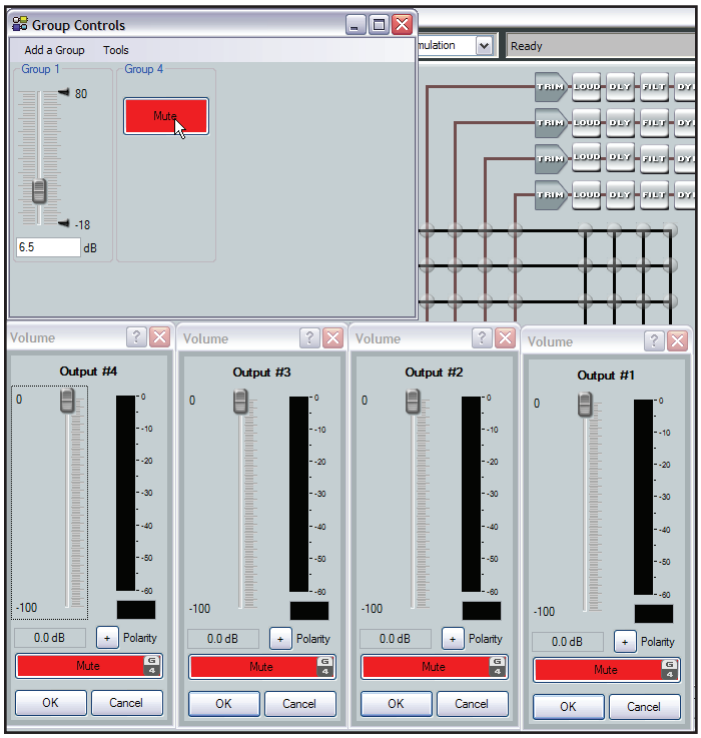


Figure 52. Sample Mute Group Master and Muted Outputs

Configuring a Group Master

To configure a group:

1. Click **Tools** > **Configure Groups** (see figure 52 on the previous page) to open the Configure Groups dialog.

   or click **View** > **Group Controls** > **Add a Group**.

2. Click the **Select Group** drop-down list (see figure 53). The list defaults to the first empty group. Select an empty group, or an existing group to overwrite.

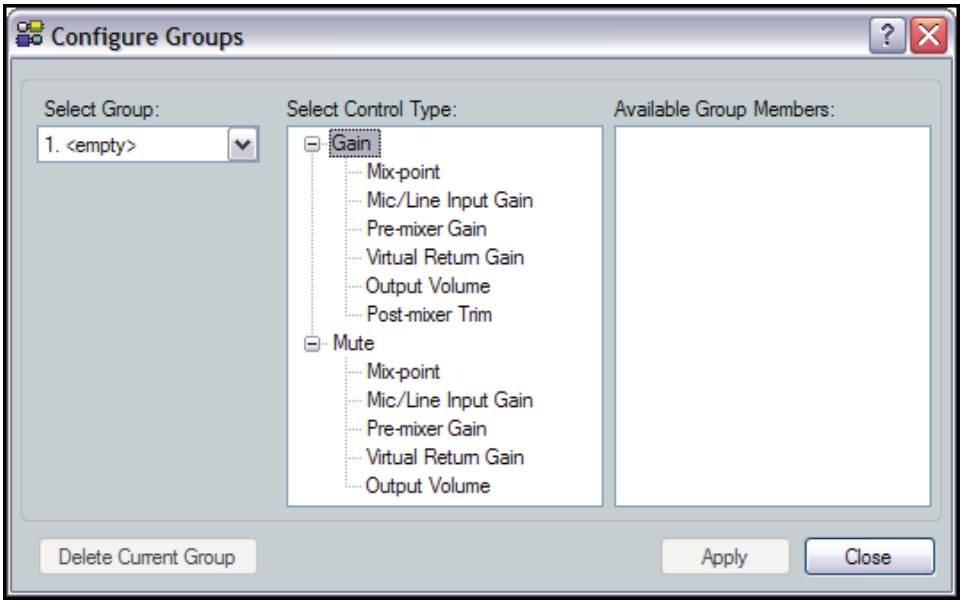


Figure 53. Configure Groups, Add Group Dialog Box

> **NOTE:** <empty> groups have no group members assigned. Numbered groups (such as <Group #1>) have controls assigned that may be overwritten if selected.

3. In the **Select Control Type** panel, expand the tree for the type of control, **Gain** or **Mute**, then select the desired control type. When a selection is made, the **Available Group Members** section populates with all possible members for the selected control type.

> **NOTE:** Potential group members in step 4 that are already assigned to a different group are displayed in **blue**.

4. In the **Available Group Members** section, click the checkbox to make appropriate selections. When a plus sign (+) exists, click to expand the tree and select individual controls. Up to 16 group members can be added.
5. Click **Apply** to create or configure the group.
6. Repeat steps 2 through 5 to create or configure up to 32 groups.
7. Click **Close** to exit the configure groups dialog.

Deleting a Group Master

To delete a group:

1. Click **Tools** > **Configure Groups** (see figure 54) to open the configure groups dialog box

   or click **View** > **Group Controls** and then click **Add a Group**.

2. In the **Select Group** drop-down list, click a numbered group (such as **Group #1**) to select it.
3. Click **Delete Current Group** in the lower left area.
4. Click **Yes** in the **Confirm Deletion** dialog box.

Viewing and Using a Group Master

Click **View** > **Group Controls** to open the group controls dialog box (see figure 54). The group controls dialog contains two menu items:

- **Add a Group** allows you to add additional groups.
- **Tools** enable you to perform various functions from the group controls dialog box.

In addition, once groups are created, a single mute button or a group fader plus the current setting readout and any soft limits that are set are visible.

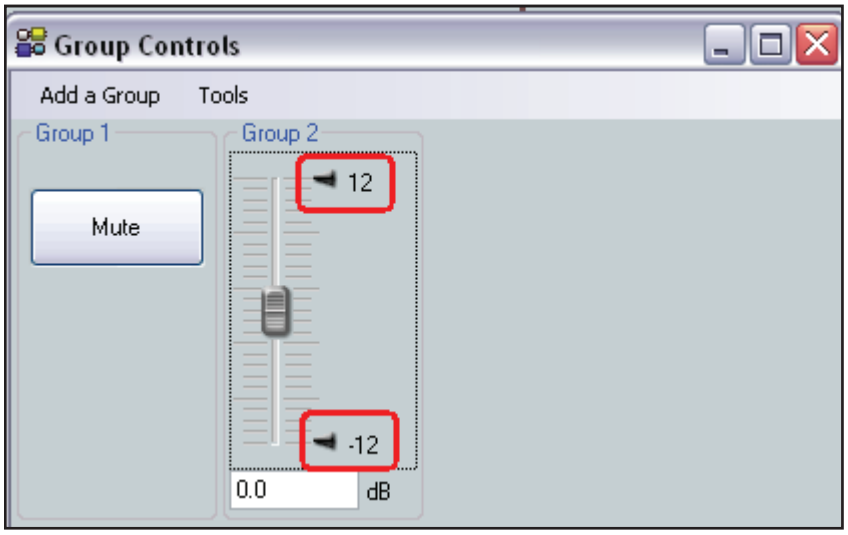


Figure 54. Group Controls Dialog Box

The group fader controls function as follows:

- Slide a group fader up and down to adjust all gain controls in the group.
- Click and drag a soft limit (◄) to set the ceiling and floor for the group.

NOTE: Soft limits cannot be dragged beyond the current setting of the group fader.

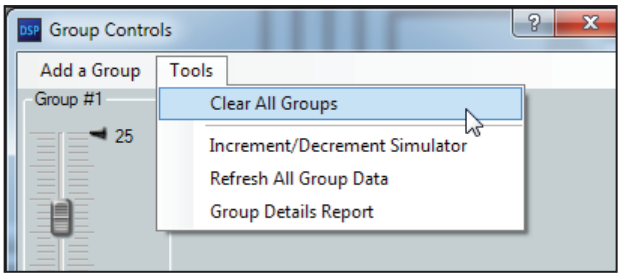
Add a Group

To launch the configure groups dialog box from group controls, click **Add a Group**. When a new group is added and the Add New Group dialog is closed, the group controls dialog box refreshes to display the added control.

NOTE: If a block is muted, that block remains muted when the group mute is released.

Tools

The Tools menu (see right) contains these selections:



- **Clear All Groups** — clears all group members and group master parameters. Soft limits are also cleared.
- **Increment/Decrement Simulator** — allows the user to test increment/decrement values (see below for more information)
- **Refresh All Group Data** — Updates group members and group master parameters.
- **Group Details Report** — generates a report, listing all group masters and membership (see **Group Details Report** on page 87).

Increment/Decrement Simulator

The Increment/Decrement Simulator provides a control for increment and decrement, with the ability to set increment and decrement values. The control is temporary, since the value is not remembered in the device.

To use the Increment/Decrement Simulator:

1. Select **Tools > Increment/Decrement Simulator**.
2. Select the group to be controlled from the **Select Group** drop-down list. The following dialog box opens:

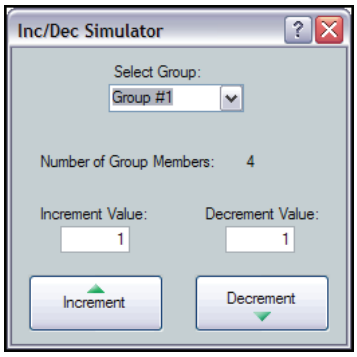


Figure 55. Increment/Decrement Simulator Dialog Box

> **NOTE:** The **Number of Group Members:** readout indicates the number of controls affected.

3. Enter an increment value and a decrement value. The default value is 1.

> **NOTE:** The size of the increment can be changed by typing a value in the **Increment Value** or **Decrement Value** field. Values can be as large as the maximum range of the control or as fine as 0.1 dB. For groups controlling mute, 1 is the only valid value.

4. Click **Increment** and **Decrement** as needed. The group master control increases or decreases by the set value to the top or bottom of its soft limit range.

> **NOTE:** When set, soft limits cannot be exceeded.

Group Details Report

Select **Tools > Group Details Report** to create a text file that details all created groups (see figure 56).

GROUP DETAILS REPORT

Group #1

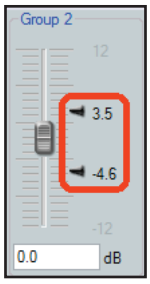
Processor Type: Output Volume
Current Mute status: Unmuted
Current Group Members:
Main Amp (Output#1) Left Channel
Stage Mixer (Output#2) Right Channel
House Video (Output#3) Left Channel
Prgm Record (Output#4) Right Channel

Group #2

Processor Type: Pre-mixer Trim
Current Gain value: 2 dB
Current Group Members:
Mic #1 (Input#1)
Mic #2 (Input#2)
Mic #3 (Input#3)
Mic #4 (Input#4)
Mic #5 (Input#5)
Mic #6 (Input#6)

Figure 56. Sample Group Details Report

Soft Limits



Each gain type control provides upper and lower soft limits used to limit the range of the group master control. Soft limits ( ), shown at left, prevent group controls from exceeding an upper limit or going below a lower limit. They are easily adjustable and provide the ability to set a ceiling and floor for the group. When a group master is created, the soft limits default to the hard limits (maximum and minimum) of that group of controls.

Soft limits can be defined using the mouse by clicking on, then dragging the **Soft Limit** icon. The resolution is 0.1 dB.
For more precise setting use the keyboard as follows:

Click within the group master fader to bring focus, then use the following key combinations:

To move the upper limit:

- <Shift + Up/down arrow> moves in 0.1 dB increments.
- <Shift + Page Up/Page Down> moves in 10 dB increments.
- <Shift + Home> moves limit to upper default. <Shift + End> moves limit to the current fader position.

To move the lower limit:

- <Ctrl + Up/down arrow> moves in 0.1 dB increments.
- <Ctrl + Page Up/Page Down> moves in 10 dB increments.
- <Ctrl + Home> moves limit to the current fader position. <Ctrl + End> moves limit to lower default.

Digital I/O Ports

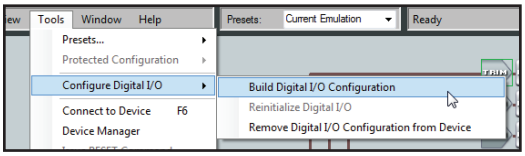
The DMP 128 provides twenty digital I/O ports that can trigger external events from DMP 128 actions, or allow external events to trigger DMP 128 actions. DSP Configurator provides pre-configured scripts with a fixed set of common trigger or event combinations. When selected, the script is compiled and placed onto the File Management system of the device. For more advanced or custom scripts, contact an Extron Electronics Applications Engineer.

When there are no scripts active, the digital I/O ports default to DI (digital input) and inactive ('Logic Hi' ≈ +5 VDC). The DI detects a Logic Hi as +5 VDC and Logic Low (active) as less than +1 VDC.

A DO (digital output) sends a Logic Lo as less than +1 VDC and a Logic Hi as +5 VDC. For every script that involves a DO, two versions are available to provide either a Logic Hi or a Logic Lo response to any action. The alternate script is designated as "Reverse DO".

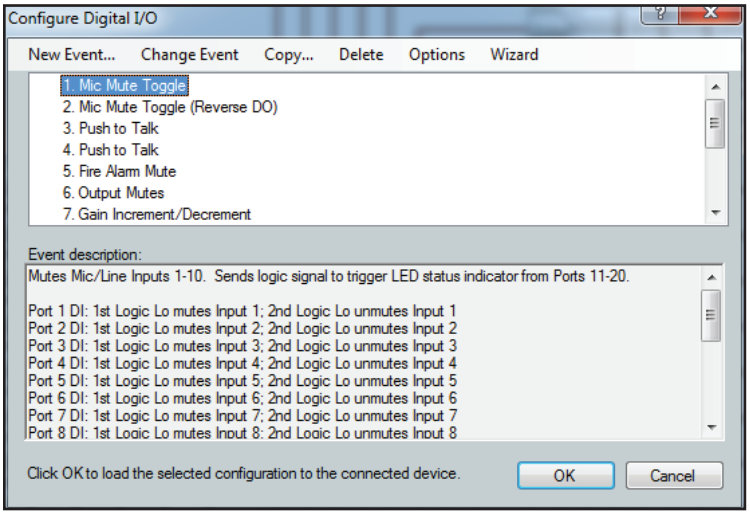
To build a script and place it into the DMP 128 File Management system:

1. From the main menu, click **`Tools>Configure Digital I/O>Build Digital I/O Configuration`**.



A dialog box opens with a list of pre-configured scripts.

2. Select a script from the **`Select a Digital I/O Configuration`** section. The `Event Description` panel describes the script and how the digital I/O ports act while the script is running. Highlight the desired script, then click **`OK`**.


Configure Digital I/O
New Event...
Change Event
Copy...
Delete
Options
Wizard
1. Mic Mute Toggle
2. Mic Mute Toggle (Reverse DO)
3. Push to Talk
4. Push to Talk
5. Fire Alarm Mute
6. Output Mutes
7. Gain Increment/Decrement
Event description:
Mutes Mic/Line Inputs 1-10. Sends logic signal to trigger LED status indicator from Ports 11-20.
Port 1 DI: 1st Logic Lo mutes Input 1; 2nd Logic Lo unmutes Input 1
Port 2 DI: 1st Logic Lo mutes Input 2; 2nd Logic Lo unmutes Input 2
Port 3 DI: 1st Logic Lo mutes Input 3; 2nd Logic Lo unmutes Input 3
Port 4 DI: 1st Logic Lo mutes Input 4; 2nd Logic Lo unmutes Input 4
Port 5 DI: 1st Logic Lo mutes Input 5; 2nd Logic Lo unmutes Input 5
Port 6 DI: 1st Logic Lo mutes Input 6; 2nd Logic Lo unmutes Input 6
Port 7 DI: 1st Logic Lo mutes Input 7; 2nd Logic Lo unmutes Input 7
Port 8 DI: 1st Logic Lo mutes Input 8; 2nd Logic Lo unmutes Input 8
Click OK to load the selected configuration to the connected device.
OK
Cancel

A dialog box opens, verifying the file has successfully uploaded to the device.

NOTE: When performing this procedure in Emulate mode, the connection dialog appears between step 3 and step 4. The DSP Configurator connects and then disconnects during the procedure, returning to Emulate mode when completed.

Reinitialize Digital I/O

Should the script stop running for any reason, select **Tools > Configure DigitalI/O > Reinitialize Digital I/O**. This option is only available in live mode.

To remove a digital I/O script from the DMP 128:

Only one digital I/O configuration can be active at a time. If the I/O activity requires modification, remove the current configuration by:

1. From the main menu, select **Tools>Configure Digital I/O > Remove Digital I/O Configuration from the Device** and click **OK**.
2. If the DSP Configurator is connected to a device, the I/O configuration is removed. If it is not connected, a connection dialog box appears.
3. Make certain the connection information is correct, then click **OK**. The I/O configuration script is removed and a confirmation dialog box appears.

Emulate Mode and Live Mode

DSP Configurator has two operational modes, Live and Emulate. In live mode, the program has established a connection and is synced with the DMP 128. Changes affect the device in real-time and changes in the current state of the device are reflected in DSP Configurator. In contrast, emulate mode allows the user to work offline, creating or editing configurations that do not immediately affect DMP 128 operation.

DSP Configurator always starts in Emulate mode. In Emulate mode, all functions of DSP Configurator are available without connecting to the DMP 128. The user can build a configuration from the blank screen, or open an existing file that contains the last configuration displayed plus saved presets. Settings and adjustments are saved to a configuration file on the PC. When the saved file is opened in DSP Configurator, all settings are restored as the current configuration (emulated if in Emulate mode or live if in Live mode).

Live mode can be entered at any time after program launch, either with a blank configuration, after creating a configuration, or after loading a previously saved configuration file.

In emulate mode, the current state is titled `Current Emulation`. In live mode, the current state is titled, `Current State`.

Synchronizing: Pull from or Push to the Device

When switching to live mode, either:

- **Pull** data from the device and update the program configuration. This option downloads device settings from the DMP 128 and synchronizes it with DSP Configurator overwriting the current DSP Configurator settings, or
- **Push** data from DSP Configurator to the device, overwriting settings in the DMP 128.

Live mode is also used to tailor audio settings in real time while listening to the audio output.

Selecting Live Mode and Pushing or Pulling Data

To switch from Emulate to Live mode:

1. Select the desired connection to the DMP 128 and make the proper connections.

   NOTE: Extron recommends connection via the Ethernet LAN port when using DSP Configurator.

2. Click **Live** (see figure 57, ②). The communication type selection dialog opens.

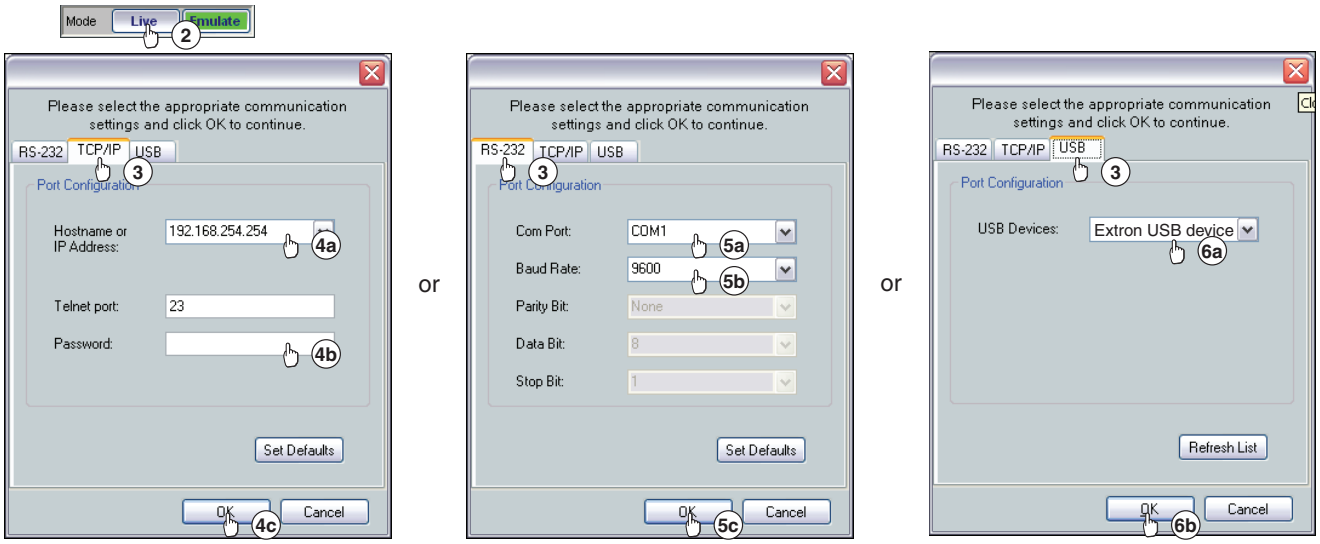


Figure 57. Selecting Live Mode

3. To select the connection method, click:
   - **TCP/IP** (for connection of the LAN port (**preferred**) — proceed to step 4
   - **RS-232** (for connection of the rear panel RS-232 port — proceed to step 5
   - **USB** (for connection of the front panel configuration port — proceed to step 6
4. If TCP/IP is selected in step 3:

   a. Observe the IP address field in the IP connection dialog box. The field displays the last IP address entered.
      - **If the IP Address field is correct**, proceed to step 4b
      - **If the address is not correct,** either click in the **IP Address** field and enter the correct IP address or click (▾) to open a drop-down list and select from among recently used addresses. Proceed to step 4b

      NOTE: If the local system administrators have not changed the value, the factory-specified default, 192.168.254.254, is the correct value for this field.

   b. If the device is password protected, click in the **Password** field and enter the appropriate administrator password

   c. Click **OK**

   The **Synchronize with Device** dialog box (see figure 58 on page 92) opens. Proceed to step 7.

5. If **RS-232** is selected in step 3:

   a. Click the **Com Port** drop-down list and select the port connected to the rear panel RS-232 port.
   b. Check the baud rate displayed in the port selection dialog box. If the baud rate does not match the device rate, click the **Baud Rate** drop-down list and select the desired baud rate. The default is 38400.
   c. Click **OK**.

   The **Synchronize with Device** dialog box (figure 58 on next page) opens. Proceed to step **7**.

6. If **USB** is selected in step 3:

   a. Click the **USB Device** drop-down list and select **DMP 128** (or **Extron USB device**, if DMP 128 is not available),
   b. Click **OK**.

   The **Synchronize with Device** dialog box (see figure 58 on next page) opens. Proceed to step 7.

7. Click either:

   a. **Pull** to configure DSP Configurator to match the device — proceed to step 9
   -or-
   b. **Push** to configure the device to match DSP Configurator — proceed to step 8

8. To push all of the gain and processor block adjustments (configuration), and all presets to the DMP 128, proceed to step 9.

   To tailor the push (push only the configuration, only the presets, or the configuration and selected presets), click **Advanced** and proceed to step 8a.

   a. Select the **Custom** radio button.

   b. Select the desired checkbox or checkboxes; **Push Configuration** and/or **Push Presets**. If **Push Configuration** is the only box checked, click **OK** and proceed to step 9.

   NOTE: **Push Configuration** includes all mix-point, gain and processor block settings. It does not include partial presets.

   c. If **Push Presets** was selected in step 8b, click **All** to select all presets or **Selected** to choose specific presets.

      - If **Selected** is clicked, click **OK** and proceed to step 8d.
      - If **All** is clicked (equivalent to a standard push), click **OK** and proceed to step 9.

   d. If **Selected** is clicked in step 8c, the **Synchronize with Device** dialog box (7b) reopens. Click **OK**. The presets dialog box opens.

   e. Select the desired partial presets to push by clicking the appropriate checkbox or checkboxes.

   f. Click **OK**. — Proceed to step 10.

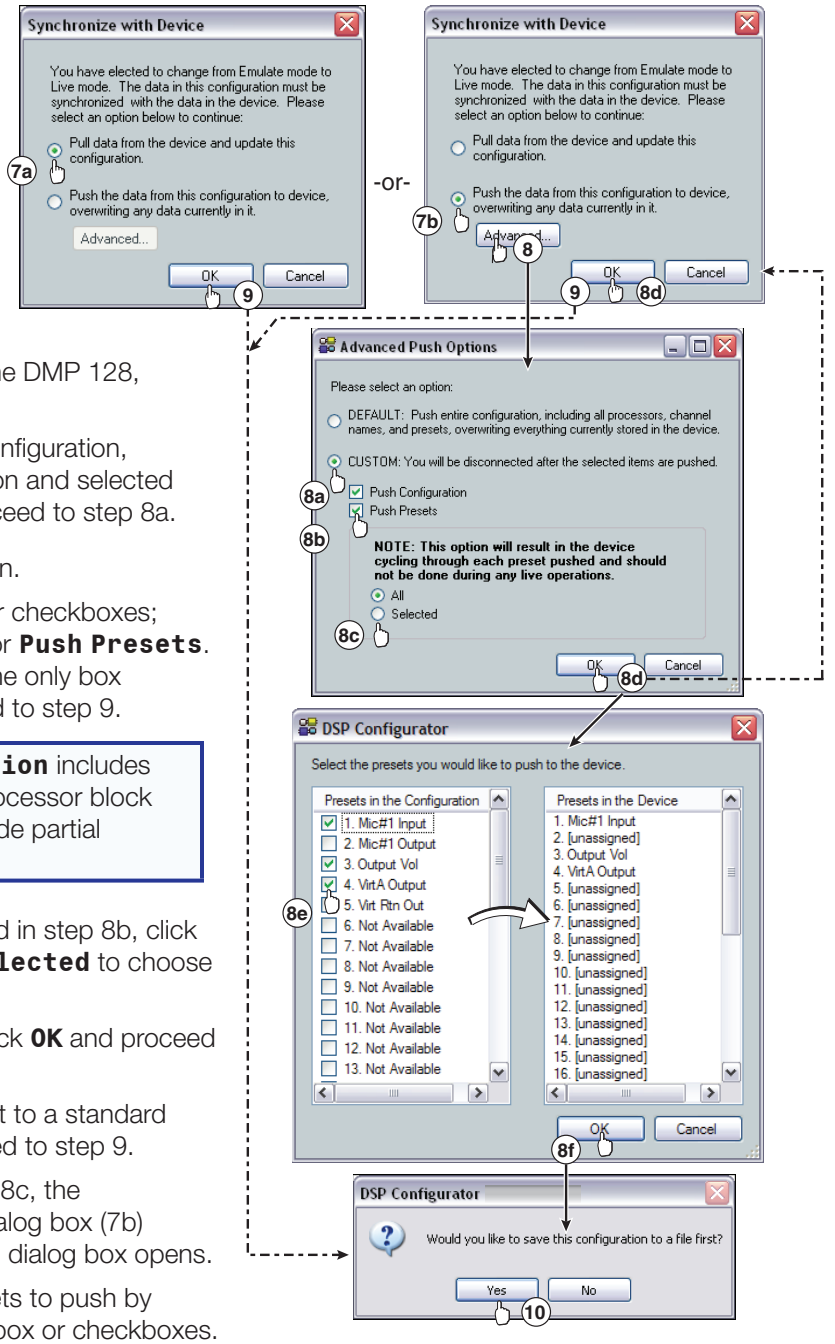


Figure 58. Selecting Live Mode (Continued)

9. Click **OK**. The DSP Configurator program is connected live to the device and the processors and presets are pushed or pulled as selected, completing the selection of **Live** mode.

10. If changes are made to the DSP parameters (including mix-point, gain or processor blocks) since the last file save, DSP Configurator prompts to save the file. Click **Yes** or **No**.

    If a password is required and not entered or if an incorrect password is entered, the program prompts for the password.

    The configuration and presets are uploaded to the DMP 128.

Presets

Presets recall a group of frequently used settings. Presets created by DSP Configurator can contain all elements (gain blocks, processor blocks, and mix-points) or a portion of the elements available within the program. In **Emulate** mode, up to 32 partial presets can be created, then uploaded as a set and stored to the device or stored to disk as a configuration file. In **Live** mode, presets can be created one at a time from the current state. They can be saved to a chosen preset number in the device, with the option to name or rename, or save to disk.

When recalled, a preset overwrites only elements contained in the preset. Presets are useful when settings for a particular room or only certain elements of a configuration need to be changed regularly.

Presets can be created in **Live** or **Emulate** modes. In **Emulate** mode, presets are created, saved to a file, then pushed to the DMP 128 when connecting in **Live** mode.

When a **pull data** synchronization method is performed, preset data remains in the DMP 128, with only the list of preset names pulled from the device. Presets in this state are marked with an asterisk until that preset is recalled (which pulls the preset data from the device), or until a backup is performed (see **Backup** on page 26). Presets pulled from the device cannot be saved to disk until they have been recalled, at which time the preset data is pulled into the DSP Configurator. Presets with no asterisk can be saved to disk.

Saved presets can be recalled via the DSP Configurator, or a control system sending an SIS preset recall command. Presets may also be saved and recalled via the embedded web page. Presets saved via the web page contain input gain, output volume, and the output mix-point settings.

Previewing and Recalling a Preset

A preset can be previewed in either **Live** or **Emulate** mode by selecting the preset from the preset drop-down list.

The program indicates a view-only preset configuration by displaying each preset element with a translucent green mask over the block.

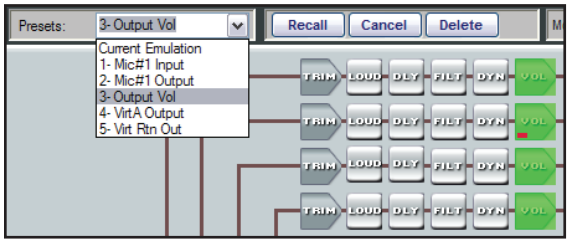


Figure 59. Preset Preview

Behavior for previewing and applying presets is as follows:

- **Live Mode** – After selecting a preset, DSP Configurator displays the preset elements that are affected by a preset recall with a translucent mask over the element, and leaves all other DSP Configurator elements unaltered. Elements without a translucent mask represent elements in the current state that are unaffected by a preset recall. Real-time changes to the current state are not reflected while previewing a preset, and the user cannot alter those elements. To apply the preset, click **Recall**. The preset reverts to **Current State**.
- **Emulate Mode** – After selecting a preset from the list, DSP Configurator displays the elements affected by a preset recall with a green translucent mask, leaving all other elements (which represent the current emulation) unaltered. Click **Recall** to apply the viewed preset to the current emulation. The preset number reverts to **Current Emulation**.

Building a Preset

Only elements of the preset highlighted (given focus) are saved as a preset. <Ctrl + A> highlights all elements within DSP Configurator.

To build a preset, highlight the desired DSP Configurator elements (gain/processor blocks, mix-points) using standard Windows keyboard and mouse actions as follows:

1. Click on the desired block to select a single block,
2. <Ctrl + click> to select multiple blocks that are not adjacent,
3. <Shift/hold + click> on the first block and click on the last block in either a vertical column or horizontal row to select multiple blocks, and
4. Click and drag a selection rectangle to select multiple adjacent blocks in either the vertical or horizontal direction.
5. Select **Tools > Presets > Mark All Items** or press <Ctrl + A>. This marks all elements within DSP Configurator, which saves a "full" preset,
6. To save the selection see **Save Preset** below.

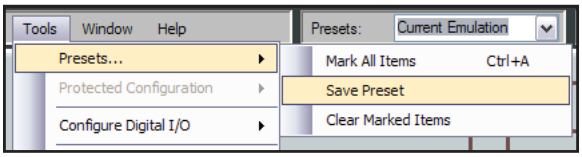
Save Preset

A preset is saved in either **Emulate** mode or **Live** mode.

Saving a preset in **Emulate** mode stores that preset in the currently open file. The DSP Configurator file is saved to disk using **File menu > Save** (recommended), and pushed to the device after a connection is established. This differs from **Live** mode where the created preset is saved in real-time to the device and becomes part of the configuration file.

To save a preset:

1. Highlight the desired preset block(s) by using click, <Ctrl + click>, <shift + click> or drag around the desired blocks.
2. Select **Tools > Presets > Save Preset**.



3. Select a preset number. In the **Preset Name** dialog box, unused presets are named **unassigned**. To create a new preset, select an unused preset number and type a preset name. If no name is entered, a default name is assigned. To overwrite an existing preset, select a preset with a name other than **unassigned**.

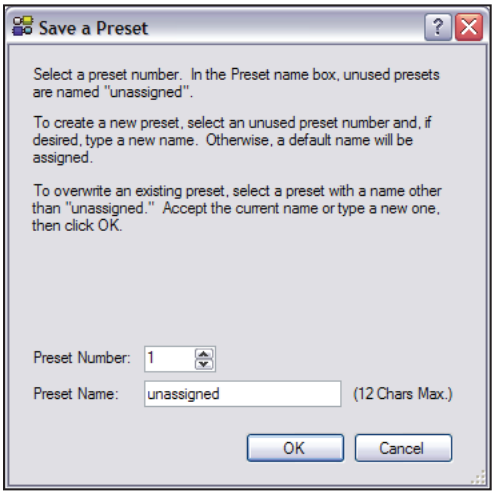
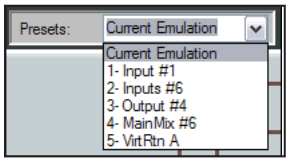

Save a Preset
Select a preset number. In the Preset name box, unused presets are named "unassigned".
To create a new preset, select an unused preset number and, if desired, type a new name. Otherwise, a default name will be assigned.
To overwrite an existing preset, select a preset with a name other than "unassigned." Accept the current name or type a new one, then click OK.
Preset Number: 1
Preset Name: unassigned
(12 Chars Max.)
OK
Cancel

Figure 60. Save Preset

4. Click **OK** to save the preset, or **Cancel** to stop the save preset operation.

Managing Presets

Once the preset is created (whether or not the DSP Configurator file is saved) it appears in the preset list, available from the DSP Configurator screen.



In **Live** and **Emulate** mode, after a preset is selected from the list, action buttons are available next to the presets bar.

The user can either **Recall** (make the preset active), **Cancel** (return to the current emulation or state) or **Delete** the preset.

In **Live** mode, select **Recall** to apply the currently displayed preset elements ("marked" elements) from the stored preset and overwrite that portion of the current state. The drop-down list switches to read **Current State**.

In **Emulate** mode, the **Recall** action button applies the currently displayed preset elements ("marked" elements) from the file and overwrites the information contained in the DSP Configurator as the current emulation, then changes the drop-down list to read **Current Emulation**.

When a preset is previewed, in either **Live** or **Emulate** mode, the **Delete** button is available. In **Live** mode, the preset is deleted from the hardware, which is reflected in software (it is removed from the preset list). After disconnecting from the device and before exiting the program, the file must be saved to retain this change. In **Emulate** mode, the preset is deleted from the file in software, which must then be saved (before exiting) to retain this change. In either **Live** or **Emulate** mode, **Cancel** defeats the preview action and returns the user to current state or current emulation, respectively.

Presets: Pull, Push, or Create Live

When a preset is pulled from the device, the preset data remains in the device until the preset is recalled. DSP Configurator pulls the names of the presets only. These presets cannot be saved to disk until recalled.

An asterisk next to the preset name indicates that only the preset name has been pulled from the device, and the preset data exists only in the device (it has not been recalled). Presets pushed to the device or created in DSP Configurator in live mode, have no asterisk. Presets with no asterisk can be saved to disk.

Protected Configuration

A protected configuration is secured with PIN protection. The protected configuration can be recalled by any user, but can only be written or overwritten using the assigned (up to) 4-digit PIN. Utilities for save/recall/change PIN, separate from preset save, are accessed from the tools menu as three sub-menus under a protected configuration menu item.

Protected configuration menu items are available only in **Live** mode from the **Tools>Protected Configuration** menu. These functions can only be performed in **Live** mode, and are unavailable in **Emulate** mode.

- Save
- Recall
- Change PIN

Save Protected Configuration

The default PIN is **ØØØØ**. The user enters the default PIN or can use the Change PIN (see below) dialog to create a new one.

Recall Protected Configuration

The dialog prompts the user to continue. Click **OK** to continue or **Cancel** the operation.

Change PIN

The change PIN utility changes a current protected configuration PIN. The current PIN must be entered before changes are allowed.

DSP Configurator Windows Menus

Keyboard Navigation

DSP Configurator is fully navigable using the computer keyboard. Some keyboard navigation behavior matches Windows standards, while other behaviors are specific to DSP Configurator.

<Tab> jumps to the various sections outlined in red in figure 61, below. When the program starts, the cursor defaults to the **`Emulate`** button (①). Press <Tab> to jumps focus to the next area, then the next, in order.

Within the sections, the <navigation arrows> move one processor block or mix-point right, left, up, or down within the section.

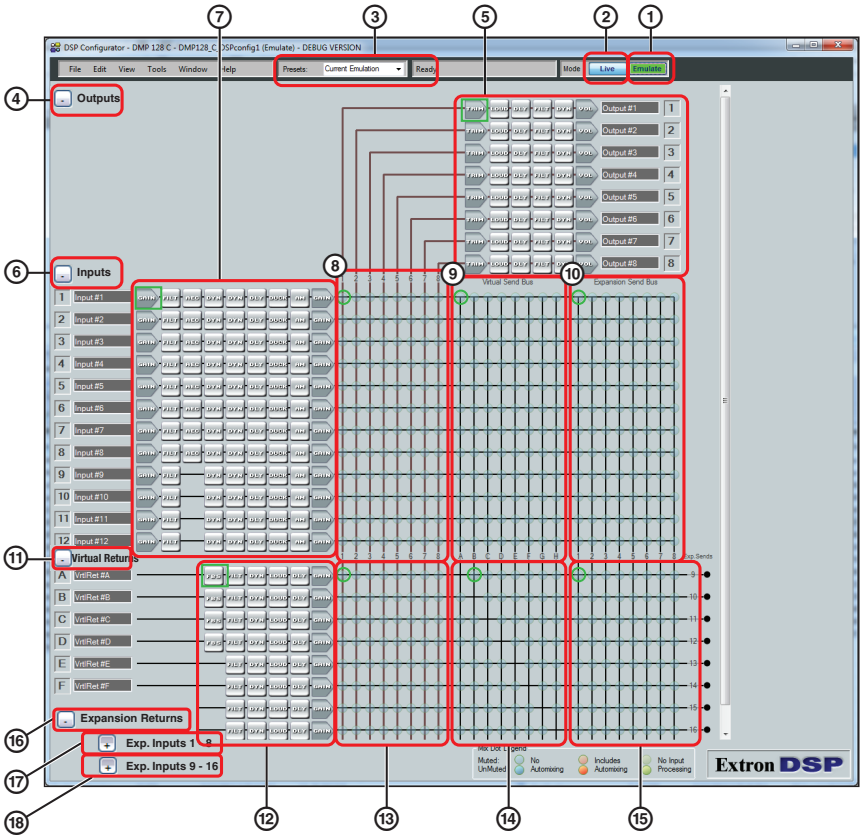


Figure 61. DSP Configurator Program

Standard Windows Navigation

The keyboard keys navigate and function as follows:

- **Tab key** —
  Sequentially jump among major segments of the DSP Configurator program. From the **`Emulate`** mode button (①), sequential jumps are in the following order:

② **`Live`** mode
③ Presets (Down arrow can select presets)
④ Outputs minimize button
⑤ Output signal path (defaults to trim)
⑥ Input minimize button
⑦ Input signal processing (defaults to gain)
⑧ Main mix-points
⑨ Virtual Send bus mix-points
⑩ Expansion Send bus mix-points
⑪ Virtual Returns minimize button
⑫ Virtual Return signal path (defaults to FBS)
⑬ Virtual Return bus mix-points
⑭ Virtual Return bus to Virtual sends
⑮ Virtual Return bus to EXP sends
⑯ EXP Returns bus minimize button
⑰ EXP Inputs bus 1-8 maximize button
⑱ EXP Inputs bus 9-16 maximize button

NOTE: The first selection in any area is always the minimize/maximize button. If the area is hidden, the next <Tab> moves the highlight to the minimize/maximize button of the next section. If the area is maximized, the next <Tab> moves to the signal processing chain or mix-points (depending on the section) before leaving the area for the next section in order.

- **Shift+Tab key combination** —
  Reverses the direction of the Tab key function.
- **Arrow (←, →, ↑, and ↓) keys** —
  Navigate up, down, left, and right within any of the areas outlined in figure 61.
- **Enter Key** — Performs the same action as a mouse double-click. For example, opens the context menu from which a processor type is selected or opens a dialog box when applicable. When an action button is highlighted, <Enter> executes the button action or toggles the button as applicable.
- **Control key** — <Ctrl> is used in the following shortcuts.
  - <Ctrl+X> — Cut the selected elements.
  - <Ctrl+C> — Copy the selected elements.
  - <Ctrl+V> — Paste the selected elements from a previous cut or copy.
  - <Ctrl+A> — The first press of the <Ctrl+A> combination highlights all A/V matrix block nodes.
- **Alt key** — <Alt> is used with specific letter keys to open and navigate task bar menus. When the <Alt> key is pressed and released, the **File** menu opens. When the Alt key is pressed and held, the first letters in the menu titles (**F**ile, **E**dit, **V**iew, **T**ools, **W**indow, or **H**elp) become underlined. Press the underlined letter key to open that menu.
- Once a task bar menu is open, use the up and down arrow keys to move up and down in the menu or submenu, use the right key to open a submenu (if applicable), and use <Esc> to back out of an active menu or submenu.

DSP Configurator-unique Navigation

Highlighting and marking items, cutting or copying, saving a preset:

When an item within the program is selected, it is highlighted by a green boundary box. One or more highlighted items can be cut, copied, pasted, or saved as a preset. The cut, copy, and paste functions can be performed using the task bar menus (see the <Alt> key, above) or the shortcuts described on the previous page.

NOTE: When an item is cut, it is not removed from its original location until it has been pasted in its new location.

Highlight multiple elements for cut, copy, paste, or a preset as follows:

1. Use the navigation (←, →, ↑, and ↓) keys to move to the first block.
2. To highlight a block:
   a. **Press and hold** the <Shift> key, then use the navigation (←, →, ↑, and ↓) keys to move away from the selected block.
   b. To highlight additional sequential blocks, **continue to hold** <Shift>, then use the navigation (←, →, ↑, and ↓) keys to move away from the selected block. Additional blocks are highlighted as long as <Shift> is held down. When the last element is highlighted, move the highlight box one additional block, then release the <Shift> key.

3. To move away from the highlighted block or set of sequential blocks, or to highlight non-sequential blocks:

   a. After highlighting blocks in step 2, press and **hold** <Ctrl>, then use the navigation (←, →, ↑, and ↓) keys to move to the next desired element. As long as <Ctrl> is held down, the block moved away from is not highlighted. If the block is highlighted, it will be unhightlighted.
   b. Release <Ctrl>, but do not press any navigation keys.

4. To highlight another element or group of elements, repeat steps 2 and 3 as required.
5. **To cut or copy**, press <Ctrl+X> or <Ctrl+C>.
6. **To save a preset**, press <Alt+T>, then right arrow →, then down arrow ↓, then press <Enter> (see figure 62).
7. The **Save a Preset** dialog box appears.

   a. <Tab> to the preset number field and type a specific preset number.
   b. <Tab> to the preset name field and type a preset name.

   NOTE: Unless entering a specific number and name, DSP Configurator enters the next sequential unused preset number.

   c. <Tab> to highlight **OK** and press <Enter>.

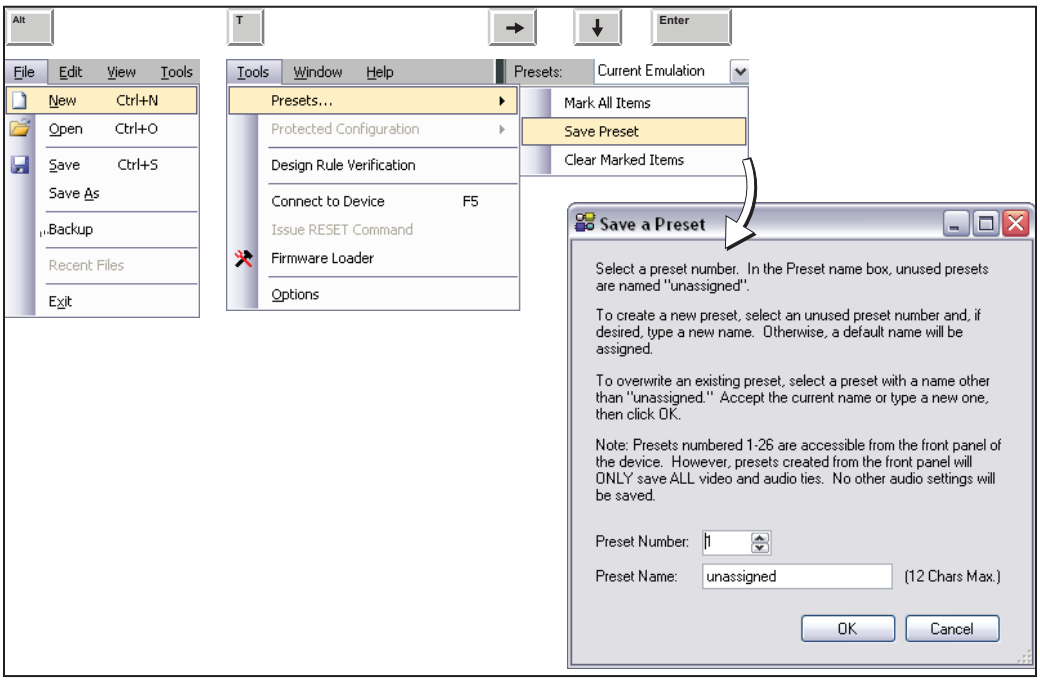

Alt
T
Enter
File Edit View Tools
New Ctrl+N
Open Ctrl+O
Save Ctrl+S
Save As
Backup
Recent Files
Exit
Tools Window Help
Presets...
Protected Configuration
Design Rule Verification
Connect to Device F5
Issue RESET Command
Firmware Loader
Options
Presets: Current Emulation
Mark All Items
Save Preset
Clear Marked Items
Save a Preset
Select a preset number. In the Preset name box, unused presets are named "unassigned".
To create a new preset, select an unused preset number and, if desired, type a new name. Otherwise, a default name will be assigned.
To overwrite an existing preset, select a preset with a name other than "unassigned." Accept the current name or type a new one, then click OK.
Note: Presets numbered 1-26 are accessible from the front panel of the device. However, presets created from the front panel will ONLY save ALL video and audio ties. No other audio settings will be saved.
Preset Number: 1
Preset Name: unassigned
(12 Chars Max.)
OK
Cancel

Figure 62. Saving a Preset Using Keyboard Navigation

Optimizing Audio Levels

The DMP 128 uses floating point DSP technology, processing data using a combination of 32- and 128-bit algorithms. The analog to digital converters (ADC) and digital to analog converters (DAC) sample at 48kHz, with 24-bit resolution.

With floating point DSP it is extremely difficult to clip the audio signal within the DSP audio signal chain, after the ADC input and before the DAC output. That means the audio signal must not be clipped at the input ADC. Clipping gives audibly undesirable results and once the audio is clipped at the input, there is no way to correct it further down the signal chain. If audio clipping occurs at the output DAC (that is not a result of clipping at the input ADC) there are ways to address it within the DSP audio signal chain.

The DSP Configurator meters indicate clipping at a user-definable point, with the default setting at –1 dB. This means the meter indicates clipping when it reaches –1 dBFS, or 1 dB below actual clipping (0 dBFS). Setting the clipping meter below actual clipping provides a "safety net", allowing the user to reduce input gain before clipping actually occurs. This safety net can be increased or decreased by selecting **`Tools > Options > Processor Defaults > Defaults > Meter Clipping`**, and setting the **`Clip Threshold`** to a number between 0 and – 20 (dB).

> **NOTE:** When the **`Clip Threshold`** is set to 0 (dB), clipping is indicated only when clipping occurs.

Meters within DSP Configurator are peak-type meters, referenced to full scale, or 0 dBFS. For the DMP 128 outputs, 0 dBFS corresponds to +21 dBu, the maximum output level of the device. Maximum input level is +24 dBu. Gain from – 3 dB to +80 dB is applied in the analog domain, while attenuation from – 3 dB to –18 dB is applied in the digital domain. The input meters are post-ADC, while the output meters are pre-DAC.

The remainder of this section references the gain, trim and volume controls outlined in figure 63.

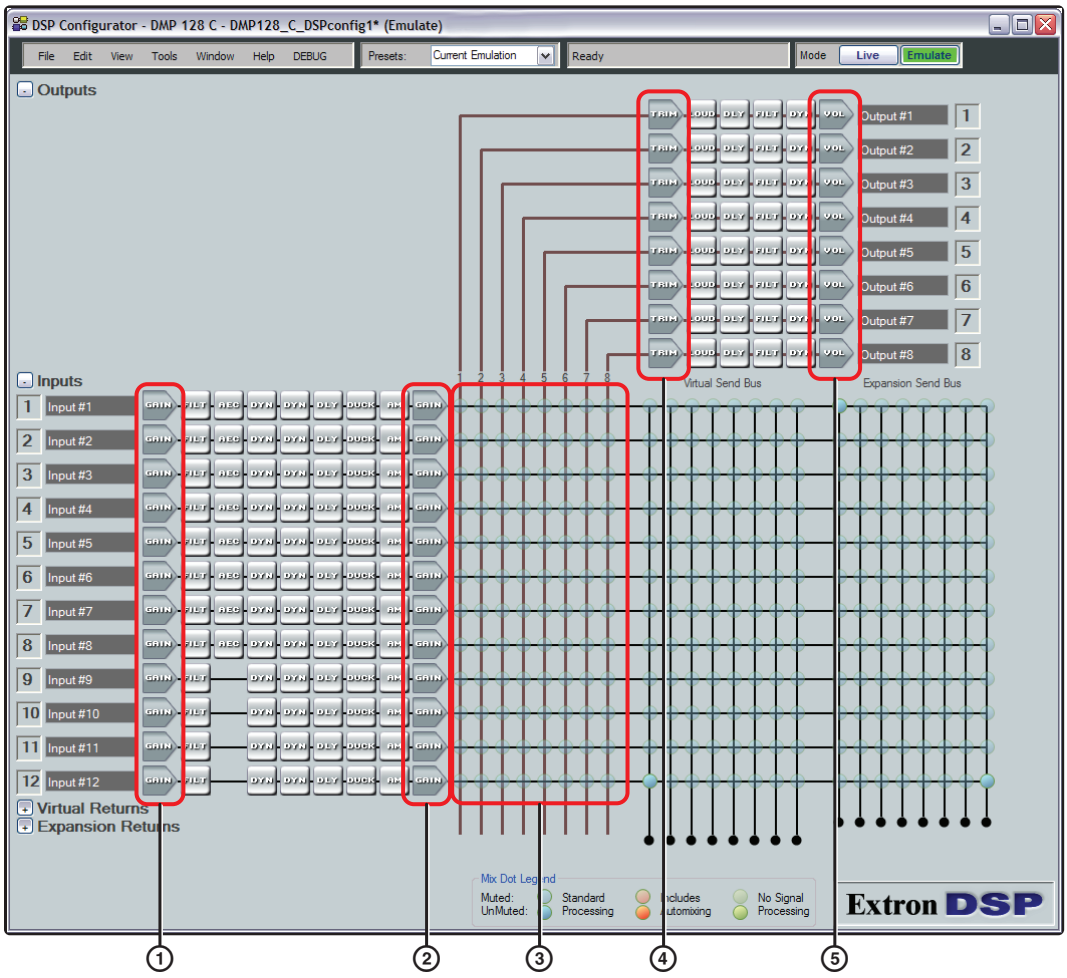


Figure 63. Gain, Trim and Volume Controls

① Mic/Line input gain
② Pre-mixer gain
③ Mix-point gain
④ Post-mixer trim
⑤ Output volume

About Setting Gain Structure

There are two approaches the system designer can take in setting up gain structure depending upon where output volume is controlled. The output volume of the DMP 128 is controlled by either of the following two gain blocks:

- Volume (⑤)
- Pre-mixer gain (②)

NOTE: While the pre-mixer gain control is not in the output signal chain, it can be used to control program level independent of mix-point levels.

In the following instructions, setup is described for output volume or pre-mixer gain when appropriate.

Setting Input Gain

Floating point DSP technology is internally more flexible than fixed point. However, the input ADCs and output DACs always run as fixed point, so it is important to optimize the audio by setting the input level as close to 0 dBFS as possible. This maintains the resolution at 24-bit. Within the DSP it is not critical to maintain audio levels at 0 dBFS in order to secure the resolution at 24-bit.

Input gain is set using the intended input source device and typical source material. When source material is not available, it can be set using pink noise obtained either from a pre-recorded track on a DVD or CD, or a signal generator.

For program material, set the input gain level (see figure 63, ① on the previous page) so the meters reach approximately – 15 to – 12 dBFS, with peaks at approximately – 5 to – 3 dBFS. This setting provides enough headroom to accommodate transients or unanticipated loud events in the program material to avoid possible clipping.

Use pink noise recorded at – 20 dBFS. If the player has an output level setting control, set the output of the player to maximum, or 0 dB of attenuation. If the maximum output setting provides gain, then back off slightly from the maximum setting. Using a signal generator, set the output at – 10 dBu. Whichever pink noise source is used, set the input gain in the DSP Configurator so the input meter reads – 20 dBFS.

Setting a Nominal Output Level

In order to set up a gain structure to include signal processing, listening to the audio can be advantageous. Route the audio that carries program material from the source to the speakers in the room. With the output volume control (see figure 63, ⑤ on the previous page) set to – 20 dB, set the external amplifier so the source material plays at a volume level that is reasonably loud but tolerable.

NOTE: Using the volume control for this purpose, set post-mixer trim (see figure 63, ④ on the previous page) to 0 dB. Using the post-mixer trim for this purpose, set volume to 0 dB (100%).

Verify the amplifier is not clipping by observing the amplifier clip indicator. This sets the amplification/volume nominal level of the system, and if desired, allows listening while making adjustments. Adjust or mute the volume control as necessary (see **Setting Volume Control for the Amplifier Stage** on page 104).

Adjusting Pre-mixer Gain

After setting input gain, add desired processors into the input signal chain. The pre-mixer gain control (see figure 63, ② on page 101) is used to compensate for level changes due to processing. Adding a compressor generally reduces the signal level, while a filter can boost or cut the overall signal level. When changes are made to filter settings after setting dynamics processors, re-check the levels in the dynamics processors to make certain they are still valid.

NOTE: This procedure is valid only when there is no active processing in the output signal path, and if the post-matrix trim value is set to 0 dB, unity gain. If processors are inserted in the output signal path, engage **Bypass** to temporarily remove them.

To adjust pre-mixer gain:

1. Open the line input gain (①), output volume (⑤), and pre-mixer gain (②) dialog boxes (see figure 63).
2. Connect program material (or pink noise) to the input,
3. Set the output volume to 100% (mute if necessary).
4. Adjust pre-mixer gain (see figure 63, ②) so the meter level of the input gain matches the meter level of the output volume. This maintains the audio at an optimal level in the input signal chain.

This sets a good starting point. After setting up the mic input gain and mix-point levels, output processing, and trim levels, if more headroom is required to prevent clipping at the outputs, return to the pre-mixer gain controls (see figure 63, ②) and lower each one by specific amounts. Further minor adjustments to the pre-mixer gain controls help to balance perceived audio levels of the different inputs.

Using the pre-mixer gain for output volume control, the procedure can be reversed. Set pre-mixer gain to 0 dB. With program material (or pink noise) present at the input, adjust the output volume until the output volume meter level is below clipping (or ideally, matches the level at the input gain meter).

Setting Output Gain Structure

Add all desired processors into the output signal chain. Keep in mind that a filter can boost or cut the overall signal level and adding a compressor generally reduces the signal level. Inserting either or both can require resetting of the output volume.

Since a limiter is the most likely choice for output processing and can only reduce the signal to prevent overload, a reduction of output level does not have to be considered. Loudness boosts the overall signal level, but only at lower volume settings

After adding processors to the output signal chain, the output volume level can clip when set to 100% (or less). Floating point DSP can overcome clipping by lowering the output volume (see figure 63, ⑤). Unless a user is prevented from changing the volume setting to 100% (or to any position where clipping occurs), it is best to adjust the pre-mixer gain (②) or post-mixer trim (④) control (see figure 63) to prevent any possible clipping.

Alternately, use the post-mixer trim controls to adjust output volume. Post-mixer trim controls provide 12 dB of gain, so use a group master with soft limits to control levels, setting an upper limit of 0 dB or less. Mic levels can also contribute to clipping at the outputs, and can need lowered to maintain the balance between program material (line outputs) and voice.

Setting Mic/Line Input and Mix Levels

In this example, the mic/line input 1 signal is sent to output 1. To set the mic/line input and mix levels:

1. Connect a microphone to Input #1.
2. Double-click the mix-point (see figure 63, ③ on page 101) for mic/line input 1 to output 1 to open the dialog box for that mix-point, and unmute the mix-point to place that signal into the mix.
   The default level for the mix-point is 0 dB, or unity gain.
3. Open the input 1 **Gain** (see figure 63, ①) dialog box and set gain to 0 dB (turn on phantom power if the mic requires it), then unmute the channel.
4. While testing the mic, raise the fader level until the mic is clearly audible. The amount of gain and the meter level reading vary at this point, but as a general guideline the input gain level is adjusted to 40 to 50 dB, with the meter averaging somewhere around – 20 dBFS.

Although audio is optimized here, voice levels at mic inputs can vary greatly and further adjustment can be necessary. When the meters average around – 20 dBFS, enough headroom is available to accommodate sudden changes to voice levels.

Adjusting Trim

This is where setting gain structure becomes a balancing act. The following sections provide guidelines, but it may take a bit of going back and forth to correctly set levels for the installation. For example, output level can be controlled and kept below clipping using a compressor or limiter in the output dynamics block. However, adjusting the post-matrix trim will affect how the compressor or limiter works.

1. Apply program material (or pink noise) at the input being adjusted.
2. Open the output **Volume** (⑤) and post-matrix **Trim** (④) dialog boxes (see figure 63).
3. Set output **Volume** to 100% (mute if necessary).
4. Adjust the post-matrix trim until the meter level in the output volume dialog is below clipping (or ideally, matches the level at the input gain meter).

This maintains the audio at an optimal level in the output signal chain and prevents clipping at the output.

Setting Volume Control for the Amplifier Stage

The maximum output of the DMP 128 is +21 dBu. As an example, assume the maximum input level of a power amp is +4 dBu with its input attenuator fully open. If using the output **Volume** control (see figure 63, ⑤) of the DMP 128 to control volume levels, to ensure clipping does not occur at the amplifier, turn down the input attenuator of the power amp the equivalent of 17 dB (21 – 4 = 17). That puts the amplifier input level at –13 dB (+4 – 17 = –13). If the amplifier setting (when the output volume controls of the DMP 128 are at maximum) is too loud for the room, it may need further reduction. If it is not loud enough for the room, a more powerful amplifier may be required.

Extron recommends using the output volume or post-mixer trim control on the DMP 128 for controlling output volume. If using loudness processing on the unit, it works in conjunction with these controls.

Using the power amplifier input attenuation to control volume (using the same power amp maximum input level) set the output volume or post-mixer trim control of the DMP 128 to –17 dB. This is another way that clip points of the two devices are matched. Verify the amplifier is not clipping by observing the amplifier clip indicator.

NOTE: Using the amplifier input attenuation to control volume compromises the signal-to-noise ratio of the DMP 128, and is not recommended.

Signal Path Building Blocks

The discrete signal paths (mic/line input, virtual return input, and line output) can be individually loaded with pre-configured, modular templates called building blocks. These blocks are designed for specific microphones, source devices or speaker destinations and can greatly streamline initial configuration. The blocks are configurable and are more versatile than a global template.

The building blocks are loaded to a selected input or output by clicking the signal path label to open the building blocks dialog box.

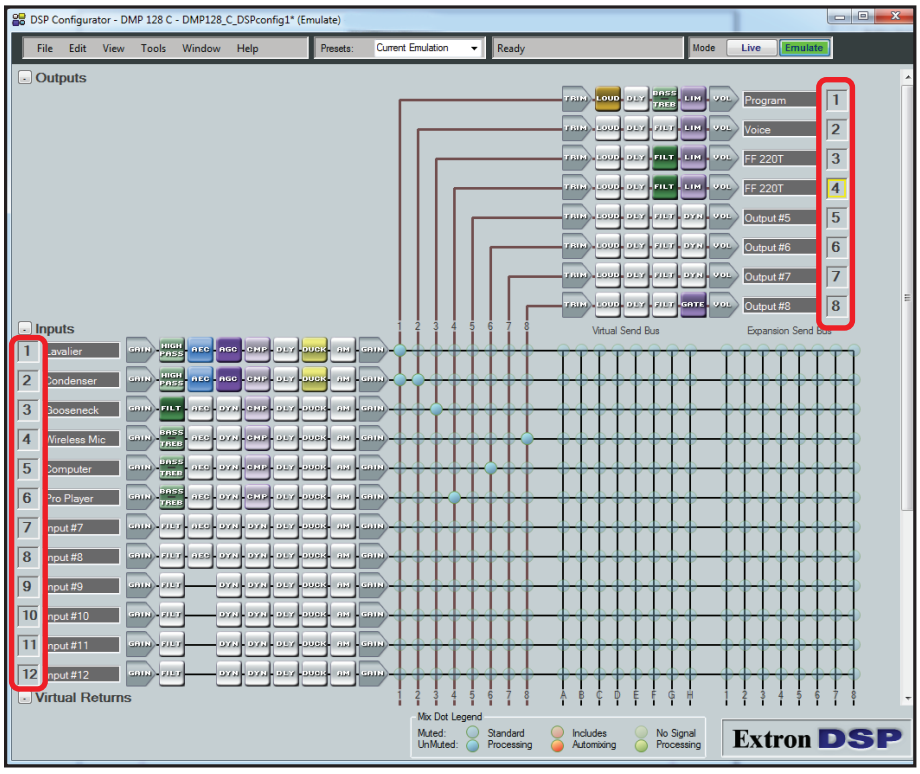


Figure 64. Building Blocks

Slightly different dialog boxes are available according to the selected signal path, mic/line input, virtual return input, or line output.

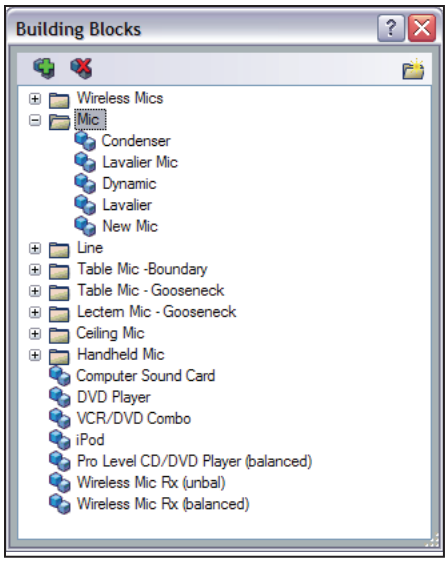
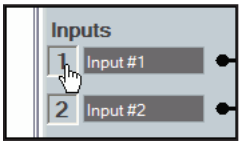


Figure 65. Building Blocks Dialog Box

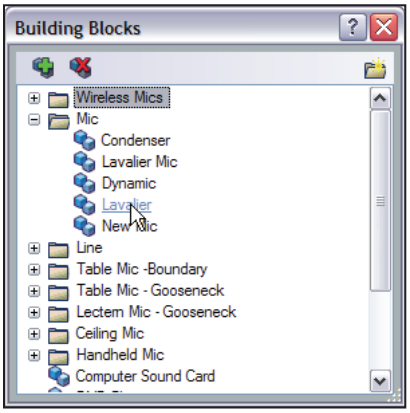
The following steps select a lavalier microphone configuration for input #1.

1. Click the **Input 1** box.



The Building Blocks dialog box (see figure 65) opens.

2. Select **Lavalier** by placing the mouse over the text **Lavalier**.



The selected text changes color and is underlined. Click the selection.

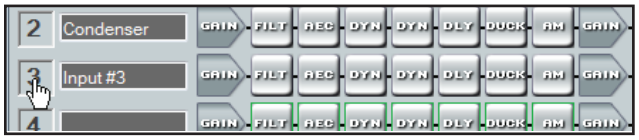
3. The input channel loads the pre-configured processor blocks, sets the gain, and names the channel **Lavalier**.

The building blocks can be renamed and processor blocks further customized according to the requirements of the system.

Adding a Building Block

Custom building blocks are created using a signal path configured for a specific device.

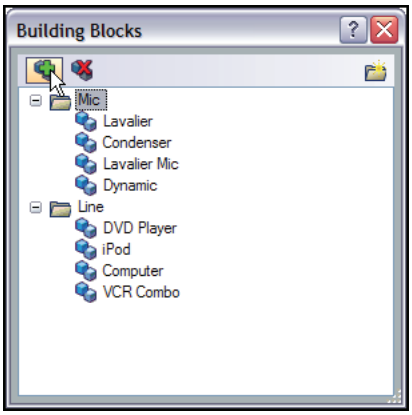
For example, when a new mic is connected to Input #3, the signal path might be tailored specifically for that mic.



In this example a gain setting is applied and a noise gate inserted. They are then stored as a custom building block.

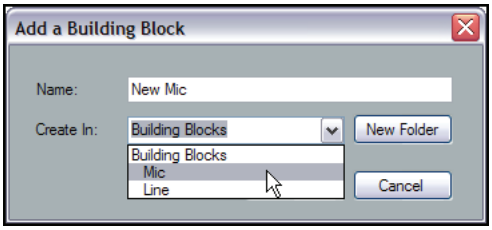
To create a building block for the new microphone:

1. Click inside the input number box (see above).
2. In the dialog box that opens, select the **Add Block** icon in the upper left.

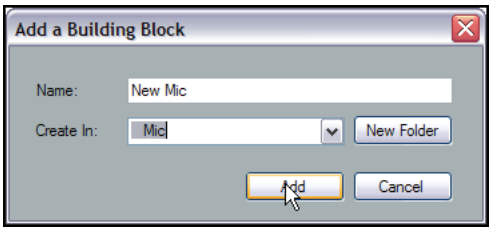


The Add a Building Block dialog box opens.

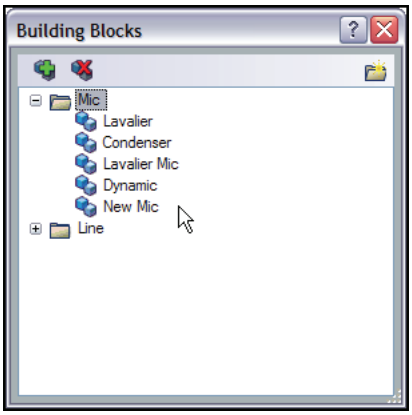
3. In the dialog box, type a name for the new device in the Name: field.



4. Select a folder for the new device (see above).
5. Click **Add**.



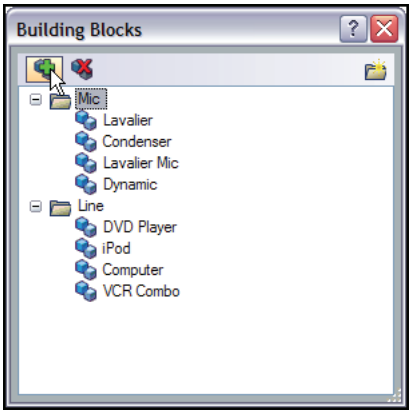
6. The new mic configuration is now a building block used to quickly configure new devices.



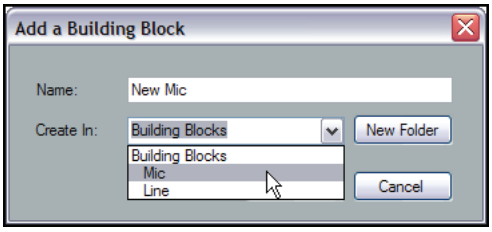
A new configuration can also overwrite existing templates.

To Overwrite an Existing Configuration:

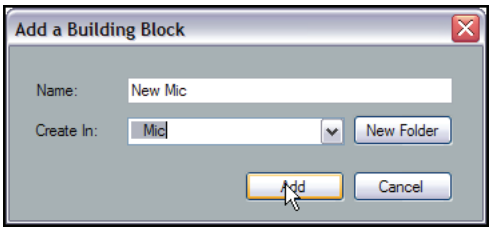
1. Click in the input number box.
2. In the dialog box that opens, select the **Add Block** icon in the upper left.



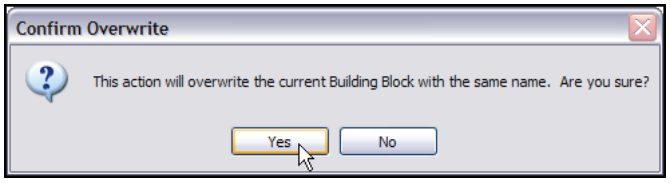
3. In the Add a Building Block dialog box, type an existing name for the new device.



4. Select a folder for the new device (see above).
5. Click **Add**.



6. A dialog box prompts to warn an existing configuration will be overwritten. Click **Yes**.



7. The new mic configuration overwrites the existing template and is ready to use.

Organize Building Blocks

The Tools menu contains a utility that allow building blocks organization and rearrangment to suit an application. Individual blocks and folders are moved or deleted and new folders created.

The general categories of folders follow the main screen of DSP Configurator and include the main inputs, virtual return inputs and the line outputs.

You can also import and export a building block file so the building blocks can be used on other computers and DMP 128 devices.

Organizing Listed Building Blocks

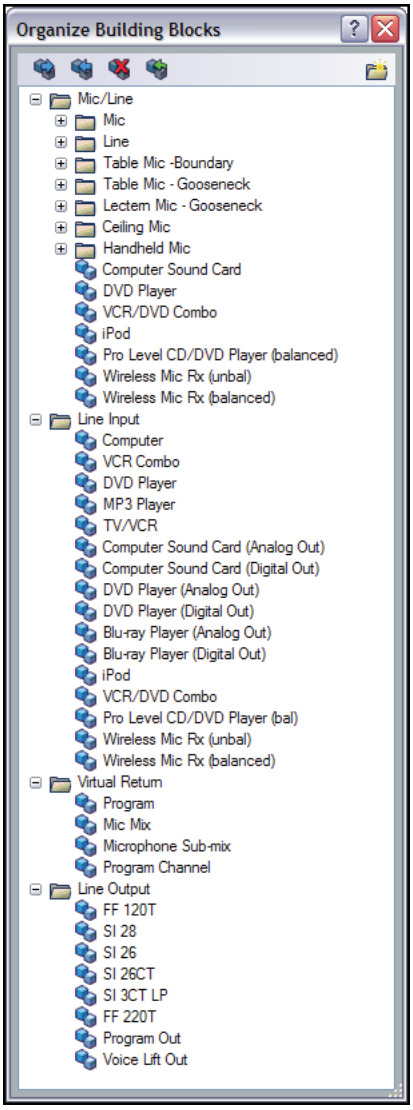
Building blocks are organized within default folders or within new folders. You can move individual building blocks or a folder with all of its contents to a new location.

To create a new folder in the Organize Building Blocks dialog box:

Click the **New Folder** icon in the upper right corner. The folder opens within the currently selected group in the organizational tree.

To move a building block or a folder, click and drag the desired item to the new location.

Click beside the folder name to expand the view of the associated building blocks.

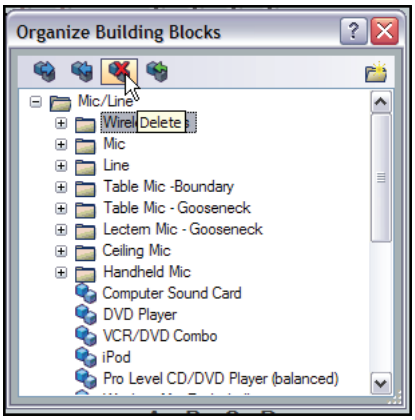


Delete a Building Block

Delete a building block to remove it from the list. If you delete a default building block, you can restore it if needed (see **Restore Default Building Blocks** below).

To delete a building block:

1. From the main menu, select **Tools>Organize Building Blocks**.
   The Organize Building Blocks dialog box opens.
2. To delete a folder and the associated building blocks, select the folder from the list and click the icon as shown on the right.
3. To delete an individual building block, select the building block and click the icon or right-click the listed building block and select **Delete** from the drop-down menu.

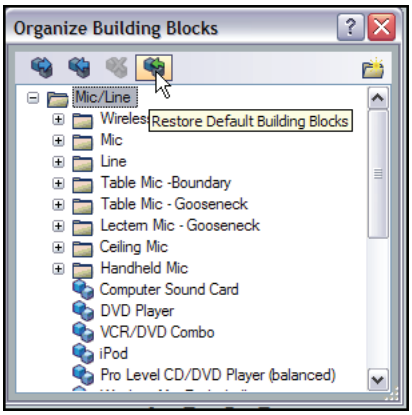


Restore Default Building Blocks

If one of the default preconfigured building blocks has been deleted, it can be restored. The default building blocks are those blocks installed with DSP Configurator. User-defined building blocks are not affected.

To restore default building blocks:

1. From the menu, select **Tools>Organize Building Blocks**. The Organize Building Blocks dialog box opens.
2. Click the **Restore Default Building Blocks** icon. The default building blocks and original folders are restored to the list.



Importing or Exporting Building Blocks

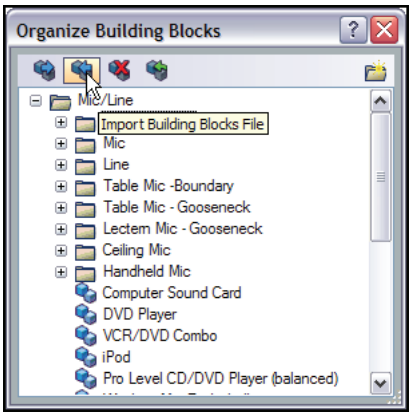
Building blocks can be imported from another computer running DSP Configurator or exported from the current computer for use elsewhere. Building blocks files are saved with an XML file extension.

To export a building blocks file:

1. From the menu, select **Tools>Organize Building Blocks**. The Organize Building Blocks dialog box opens.
2. Click the **Export Building Blocks File** icon. The "Export to..." dialog box opens.
3. Browse to the location where the file is to be saved.
4. In the **File name** field, leave the current file name or enter a new file name.
5. Click **Save**.

To import a building blocks file:

1. From the menu, select **Tools>Organize Building Blocks**. The Organize Building Blocks dialog box opens.
2. Click the **Import Building Blocks File** icon. The "Import from..." dialog box opens.
3. Browse to and select the desired building blocks file.
4. Click **Open**. The selected building blocks file is imported into the Organize Building Blocks dialog box.



Dante Installation and Operation (AT Models Only)

This section describes the DMP 128 AT network installation, configuration, and control using Dante Controller for Windows and includes:

- **DMP 128 AT Bus**
- **Dante Controller Software Installation**
- **Creating a Physical Dante Network**
- **Dante Controller Configuration**
- **Selecting Inputs and Outputs for Dante**
- **Signal Routing with Dante**
- **Using the Extron Expansion Bus with DMP 128 AT Devices**

DMP 128 AT Bus

DMP 128 devices equipped with an AT bus use Dante technology to provide high performance digital audio networking over standard TCP/IP networks. Dante allows audio channels to be transported across a switched Ethernet data network while meeting the quality requirements of professional audio.

Audio signals are converted by the DMP 128 AT models, processed, then routed to the AT bus for availablity over the network. The audio network can be shared with control traffic or unrelated data traffic. Audio channels can be unicast or multicast to make the best use of available bandwidth.

The DMP 128 AT models are both a transmitter (source), and receiver (destination) for signals on a Dante network. Transmitters and receivers can be located anywhere on the Ethernet network and can be relocated without reconfiguring the DMP 128 AT or the network.

Dante Controller Software Installation

The DMP 128P AT uses Dante Controller for Windows from Audinate to select and route inputs and outputs for all connected DMP 128 ATs and other Dante-compatible audio devices.

Download and install the latest version of Dante Controller for Windows from the included software DVD or from the Dante Controller product page at **www.extron.com**.

Dante Network Audio Setup

The following procedure is recommended for configuring DMP 128 AT devices on a Dante network. When configuring multiple DMP 128 AT models, follow this procedure for each device, connecting them to the host computer one at a time.

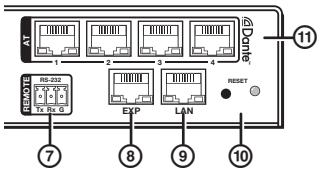


Figure 66. Rear Panel, AT Switch (11)

NOTE: A connection to both the Dante (AT) Ethernet switch and the DMP 128 LAN port is required for this setup.

- If the control computer has only one Ethernet port, connect it to a rear panel AT port (see figure 66, 11). Then connect a short Ethernet cable from the LAN port (see figure 66, 9) to an unused AT port (see figure 66, 11).
- Alternately use an Ethernet switch to connect the control computer to both the LAN and AT ports.

1. Connect the control computer to an AT (Dante) port on the DMP 128. Also connect the control computer to the DMP 128 LAN port.
2. Start DSP Configurator in **`Live`** mode (see **Selecting Live Mode and Pushing or Pulling Data** on page 90).
3. From the DSP Configurator menu bar, select **`Tools > Device Settings`** (see figure 67, right).

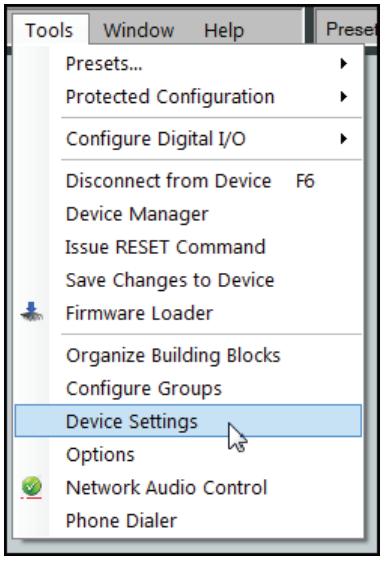


Figure 67. Tools, Device Settings

4. In the **Unit Name:** field, enter a descriptive name for the DMP 128 (typically the device name and location) to enable easy identification on the network (see figure 68).

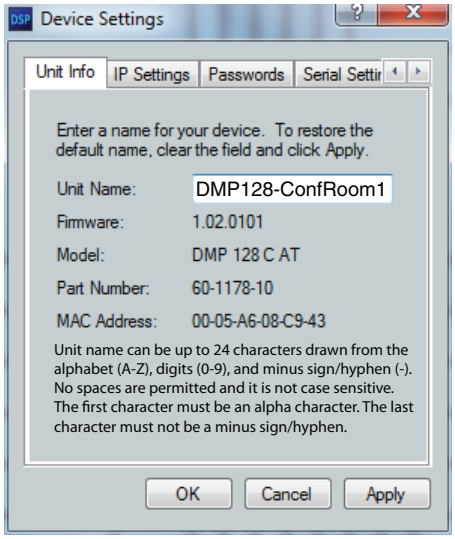

Device Settings
Unit Info
IP Settings
Passwords
Serial Settir
Enter a name for your device. To restore the default name, clear the field and click Apply.
Unit Name:
DMP128-ConfRoom1
Firmware:
1.02.0101
Model:
DMP 128 C AT
Part Number:
60-1178-10
MAC Address:
00-05-A6-08-C9-43
Unit name can be up to 24 characters drawn from the alphabet (A-Z), digits (0-9), and minus sign/hyphen (-). No spaces are permitted and it is not case sensitive. The first character must be an alpha character. The last character must not be a minus sign/hyphen.
OK
Cancel
Apply

Figure 68. Device Settings - Unit Name

5. Click **Apply**, then **OK** to close the dialog box.
6. Open the Dante Controller (see **Start Dante Controller** on page 118).
7. From the Dante Controller menu bar, select **Device > Device View** (see **Rename a Dante Device** on page 120).
8. Choose the device. The DMP 128 appears as **DMP128**, plus the last six digits of the Dante MAC address.

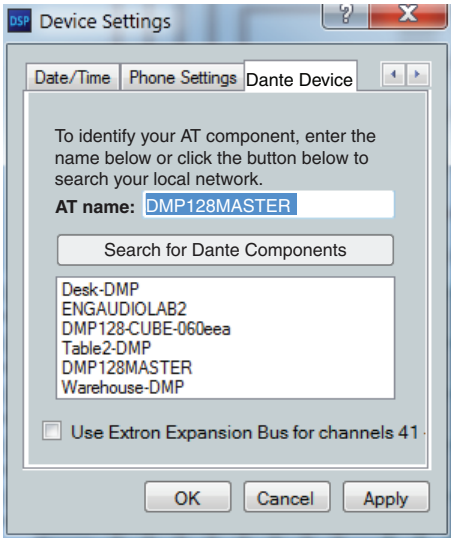
NOTE: Connect DMP 128 units to the Dante network one at a time to easily identify which one is being named. To confirm the proper device, compare the Dante MAC address printed on the rear panel of the unit with the MAC address shown in figure 68.

9. Select the **Device Config** tab in Dante Controller.
10. Rename the device to match the DSP 128 name given in DSP Configurator (see step 4 above), then press <Enter>.
11. Select **Yes** to confirm the new name.
12. To connect additional DMP 128s, repeat this procedure with the next device.

Dante Device Discovery

The Tools menu also provides a way to locate other Dante devices on your network. Once the device is named (see **Dante Network Audio Setup** on page 113), it can easily be identified as a Dante host from any other Dante device.

To locate Dante devices on an audio network:



1. Select **Tools>Device Settings**. The Device Settings dialog opens.
2. Select the **Dante Device** tab.
3. Dante devices are found by either:
   a. Entering the name in the AT name: field,
   -or-
   b. Click **Search for Dante components** to start a search.
   The search may take a few minutes.

   As new devices are discovered, they are listed in the window.
4. If you are using the EXP port to connect to a non AT DMP 128, check the **Use Extron Expansion Bus for channels 41-56** box (see **Using the Extron Expansion Bus with DMP 128 AT Devices** on page 125).
5. Click **Apply** or **OK**.

Creating a Physical Dante Network

Basic Configuration

A physical network is required to share Dante audio channels between DMP 128 AT devices. Each DMP 128 AT contains a 4-port Gigabit switch with four RJ-45 connectors located on the back panel to accept standard network cables. The 4-port switch can operate as a standard switch described in this section or in redundant mode (see **Redundant Configuration** on page 117) that switches between duplicate audio streams during a single network failure, preventing audio stream interruption.

NOTE: The Dante Controller network configuration defaults to switched mode (basic) making star and daisy-chain configurations possible. When configured for redundant operation, different connection rules apply (see **Redundant Configuration** on page 117).

A DMP 128-based Dante network can be configured in a daisy-chain or Star network topology using the four port switch and the Dante Controller in Switched mode.

Star network topology has one DMP 128P AT as the central unit, which is then directly connected to up to three other units. Alternatively, a larger network switch can be used in place of the central DMP 128 AT, allowing more than three units to be connected in the star configuration (see figure 69).

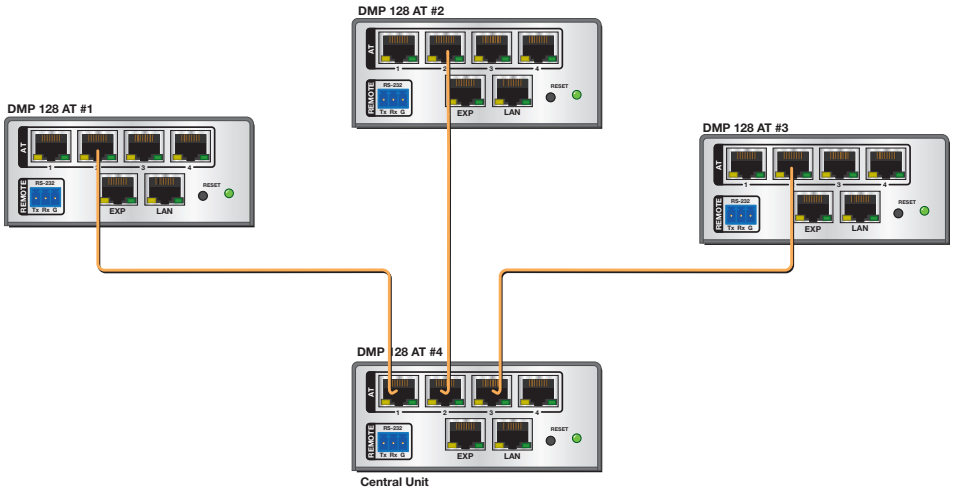


Figure 69. Star Network Topology

A daisy chain configuration can also be used. Each unit is connected to both the previous unit and the next unit in the chain (see figure 70).

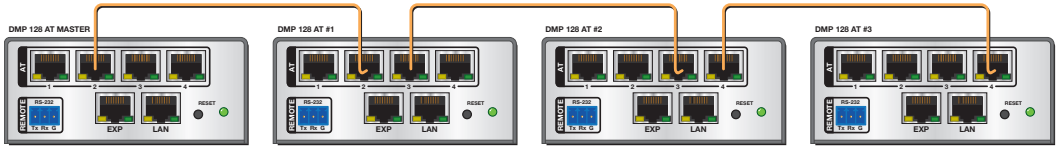


Figure 70. Daisy Chain Topology

Hybrid versions combining the star and daisy chain topologies can also be built, but a ring topology, or any topology that creates a duplicate connection causes a connection failure in the Dante Controller software. Redundant connections are possible using the primary and secondary port interface structure (see **Redundant Configuration** on page 117).

NOTE: Connections between DMP 128 AT ports in either a star or daisy chain network do not need to be sequential, nor do they need to be made between the same port numbers.

Redundant Configuration

Redundant network configurations can be employed, however, you must observe the proper order of rear panel AT port connections. In redundant mode the 4-port switch acts as two separate 2-port switches that duplicate audio traffic (see figure 71). The two sets of switches (primary and secondary) cannot be connected together anywhere in the system.

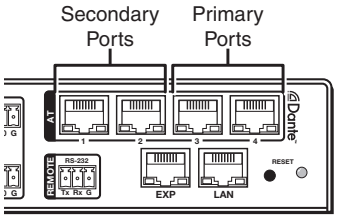
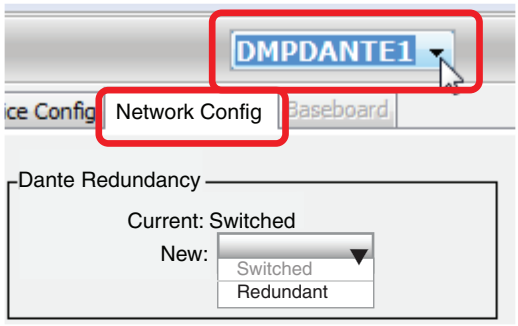


Figure 71. AT Ports—Primary and Secondary Ports Configuration

At least two separate network systems are required for redundancy. The primary ports (3 and 4) are connected to one network and the secondary (1 and 2) to another.

The redundant mode must be selected with Dante Controller (see **Start Dante Controller** on page 118).

1. Launch Dante Controller
2. Access the Device View .
3. Double-click the device name from the Routing View.
4. Select the **`Network Config`** tab.
5. In the `Dante Redundancy` section, from the `New` drop-down list, select **`Redundant`**.



The Dante Controller PC must always be connected to one of the two primary ports on the DMP 128 AT switch (see figure 71).

NOTES:

- In redundant mode, control information is carried only by primary ports 3 and 4. If a control PC is connected running Dante Controller, it must use port 3 or port 4.
- AT connections follow the same distance limitations as standard Ethernet connections.
- Potential issues of network flooding can occur. Check the Audinate website (**www.audinate.com**) for FAQs regarding managing network traffic.

Managing Network Traffic

Because Dante devices prefer to use multi-cast to distribute audio channels, it is recommended that, if possible, the audio network be kept separate from the local business network. For more information on strategies to manage audio network traffic in an integrated system, refer to the Dante Controller manual (Multicast Transmit Flow Configuration).

Dante Controller Configuration

Now that the DMP 128 ATs are configured for network operation and the Dante software is installed, Dante Controller must be configured.

Start Dante Controller

To launch Dante Controller.

From the DSP Configurator menu bar, select **`Tools>Network Audio Control`**.

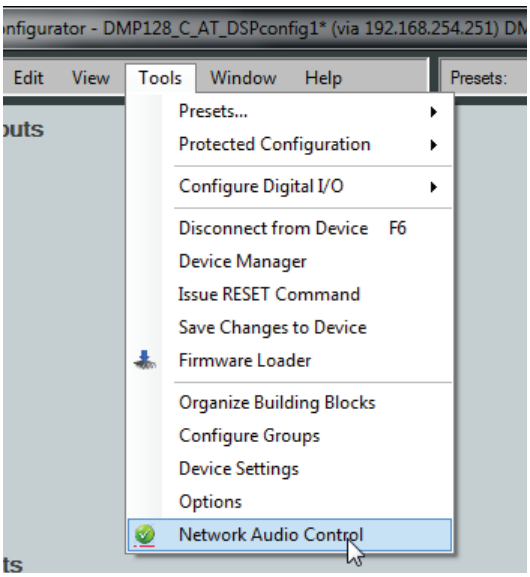


Figure 72. Network Audio Control

Dante Controller launches. If the network has a DHCP server, it receives its IP configuration using the standard DHCP protocol. On a network without a DHCP server, a Dante-enabled device automatically assigns itself an address using link local protocols.

Once connected, Dante Controller begins to auto-discover other Dante devices. While scanning the network, it also advertises itself to allow other Dante-enabled devices to see and communicate with it.

When the network scan is compete, Dante Controller opens in Network View.

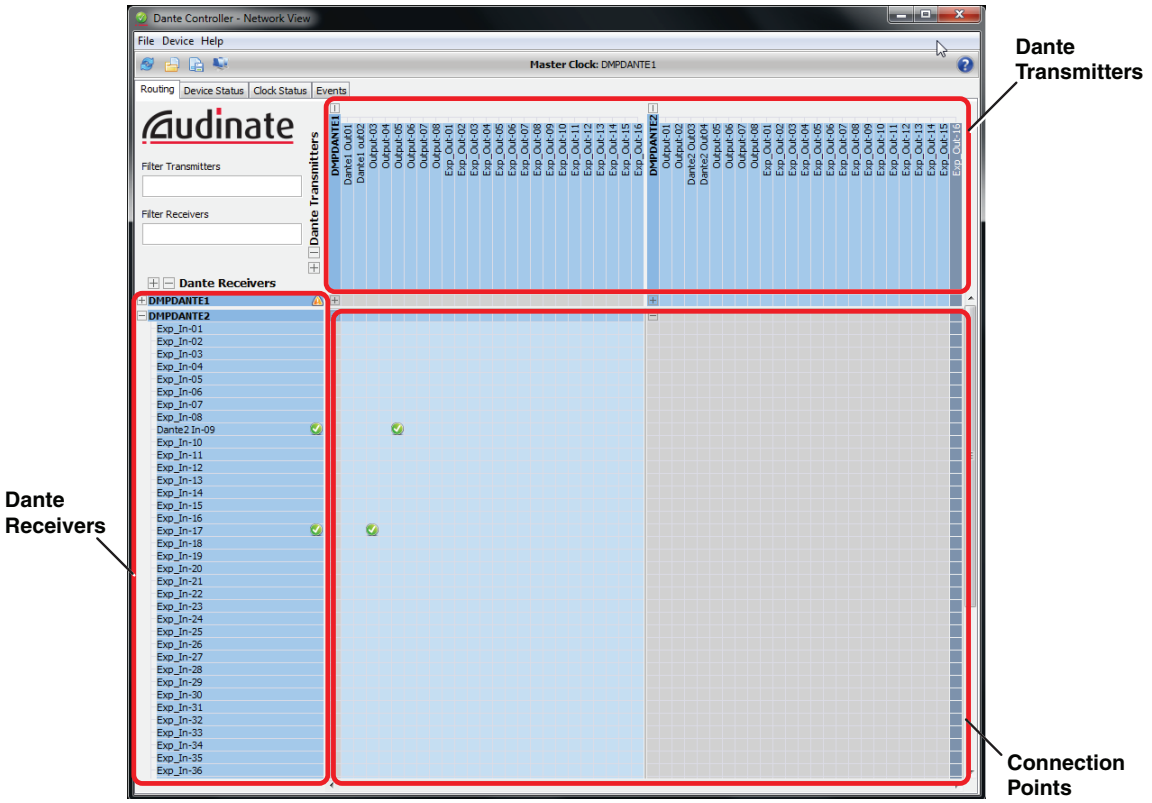


Figure 73. Dante Network View

Receivers, representing Dante device inputs, are on the left side running vertically. Transmitters, representing Dante device outputs, are along the top running horizontally. The connection points form a large matrix in the center of the screen.

The name initially shown in the first column is extracted from the Dante interface of the connected device. The DMP 128 AT expansion inputs are Dante receivers and labelled `EXP_In-Ø1` to `EXP_In-56` (or the maximum available from each connected device). The output names are labelled from `Output-Ø1` to `Output-Ø8` to mirror the DMP 128 line outputs, and from `EXP_Out-Ø1` to `EXP_Out-Ø8` that correspond to the DMP 128 expansion outputs and `EXP_Out-Ø9` to `EXP_Out-16` that correspond to the DMP 128 virtual returns 9 to 16.

Dante provides a way to rename the connected device and to re-label each input and output (Dante Controller does not automatically use the DMP 128 AT device name). Follow the setup procedure (see **Dante Network Audio Setup** on page 113) keeping both names identical or at least as similar as possible to avoid confusing device names during operation. Inputs and outputs can also be renamed as necessary before proceeding.

Dante Controller Naming Conventions:

- Device labels follow Domain Name System (DNS) hostname rules. Legal characters are `A-Z`, `a-z`, `Ø-9`, and dash or hyphen (-). Device labels must begin with a letter.
- Dante Tx and Rx labels can be up to 30 characters in length. Label comparisons are not case-sensitive; "`OUTPUTØ1`" and "`outputØ1`" are treated the same. Unicode and non-roman characters are not supported.
- Tx channel labels can use any character except equals (=), period (.), or '@'. Tx channel labels must be unique on a device but do not need to be unique on the network. On the network, channel IDs are prefaced with the device ("`channelØ1_DMP128-Ø1`", or "`channelØ1_DMP128-Ø7`").
- Rx channel labels follow the same rules as Tx channel labels.

Rename a Dante Device

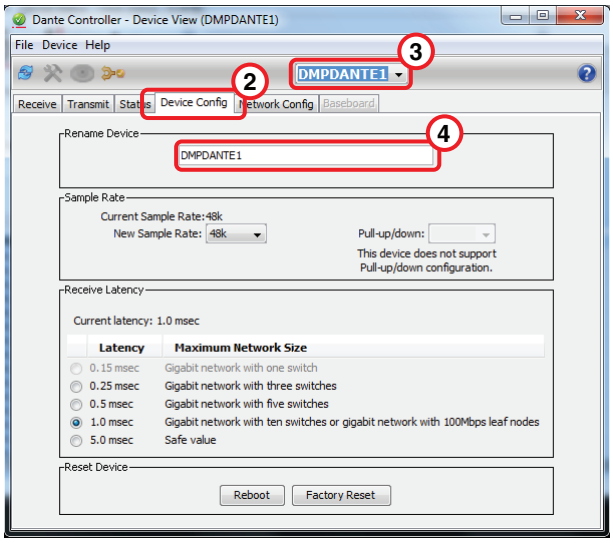


Figure 74. Dante Controller - Rename Device

To rename a connected device:

1. From the toolbar, select **Device > Device View** or use <Ctrl+D>.
2. From the Device View dialog box, select **Device Config**.
3. From the pulldown list, select the device to rename.
4. In the **Rename Device** field, type the new name for the device following the naming conventions note on the previous page.
5. Repeat steps 2 to 5 for each connected device.
6. When all devices have been renamed, from the toolbar, select **Device > Device View** to return to the network view screen.

AT Expansion Bus Organization

With 56 inputs and 24 outputs for each DMP 128 AT device, the Dante connection points can quickly become unmanageable in large systems. To better organize the various inputs and outputs, it is highly recommended to name each input and each output using descriptions of the device they belong to, the location of the device, and even the purpose of the input or output.

- Receiver channels (Inputs) 1-56 (from Dante)
- Transmitters (Output) 1-8 are the DMP 128 line outputs one to eight.
- Transmitters EXP-Out-Ø1 to EXP-Out-Ø8 are the DMP 128P AT expansion outputs.
- Transmitters EXP-Out-Ø9 to EXP-Out-16 are the DMP 128 Virtual Sends, post processing.
- Names from the Dante Controller are read by the DMP 128 and assigned to the proper input or output.

Rename an Input or Output

To simplify setup and operation of large matrix systems, rename the input (receiver) and output (transmitter) lines to a label indicating the input a source device is connected to or the output a receiver is connected to.

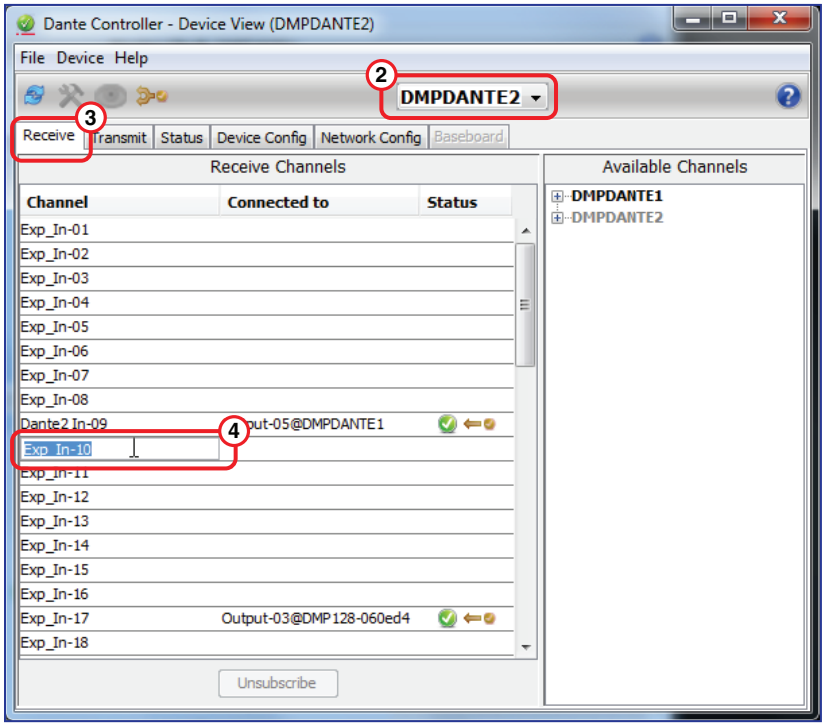


Figure 75. Dante Controller - Rename Inputs and Outputs

To rename an input:

1. From the menu, select **Device > Device View** or use <Ctrl+D>.
2. From the drop-down list in the upper center of the screen, select the device.
3. From the Device View screen, select the **Receive** tab.
4. Click inside the input text box to rename. A cursor appears.
5. Type the new name (up to 30 characters) using any combination of letters, numbers, and special characters.
6. Press <Enter>, or click in another input text box.
7. Repeat steps 4 to 6 for each input line.
8. When all Inputs have been renamed, close the dialog box.

To rename an output:

1. Follow steps 1 and 2 above.
2. From Device View, select the **Transmit** tab.
3. The transmit tab has an extra column for naming the outputs. Click inside the **Channel Label** text box to rename an output. A cursor appears.
4. Type a new name (up to 30 characters) using any combination of letters, numbers, and special characters.
5. Press <Enter> or click in another output **Channel Label** field.
6. Repeat steps 3 to 5 for each input to be renamed.
7. When all outputs have been renamed, close the dialog box.

Selecting Inputs and Outputs for Dante

Dante Controller is used for all network level signal routing, however DSP Configurator assigns the input and output channels to the AT bus. The line outputs, expansion outputs, and virtual returns are available on the AT bus outputs.

DMP 128P AT Bus Outputs

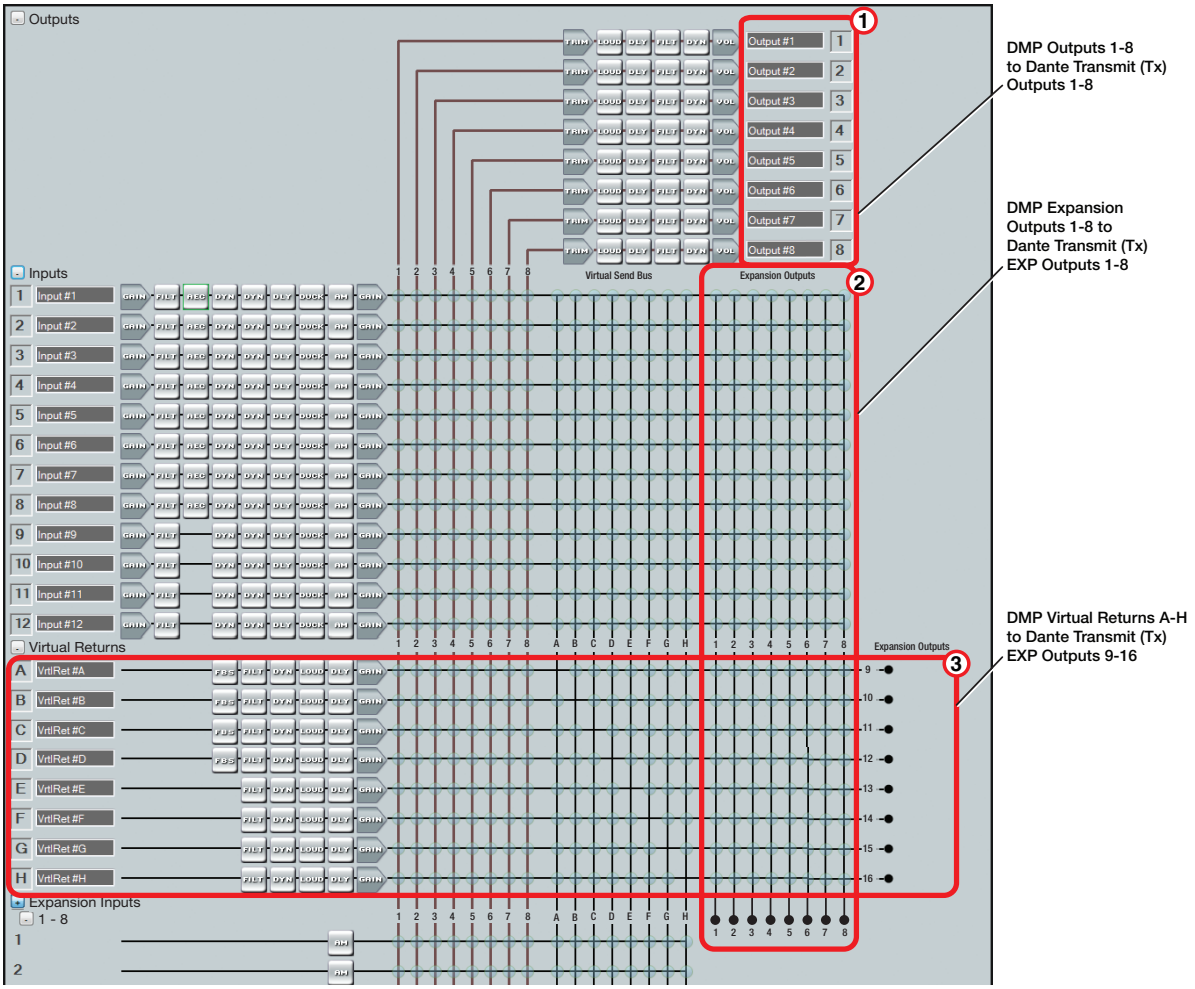


Figure 76. DSP Configurator AT Outputs (Dante Transmitter Channels)

In figure 76, the three areas that provide signal on the AT bus for the Dante transmitter channels (see figure 77, next page) are outlined. All DMP 128 AT EXP outputs (see figure 76, ①, ②, and ③) appear to Dante Controller as transmitters. The audio from these outputs is always available to Dante compatible devices on the same network, and appear as transmitters (see figure 77, ④ on the next page) in Dante Controller.

The DMP 128 output lines are always the first eight transmitter lines (Outputs 01 through 08 in figure 77, ④). The eight expansion outputs (see figure 76, ② on the previous page) appear in the Dante transmitter list (item ④, below), as EXP_Out-Ø1 to EXP_Out-Ø8). The eight virtual sends of the DMP 128 (figure 76, ③) appear as EXP_Out-Ø9 through EXP_Out-16.

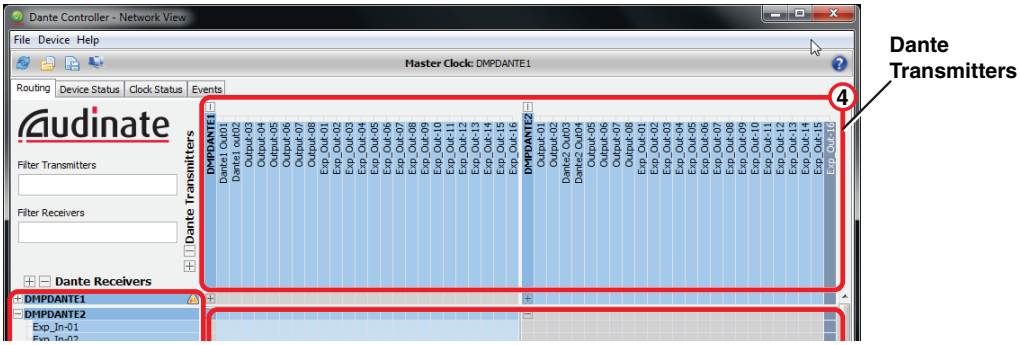


Figure 77. Dante Transmitters

DMP 128P AT Expansion Bus (AT Inputs)

There are a total of 56 DMP 128 EXP inputs that can be routed.

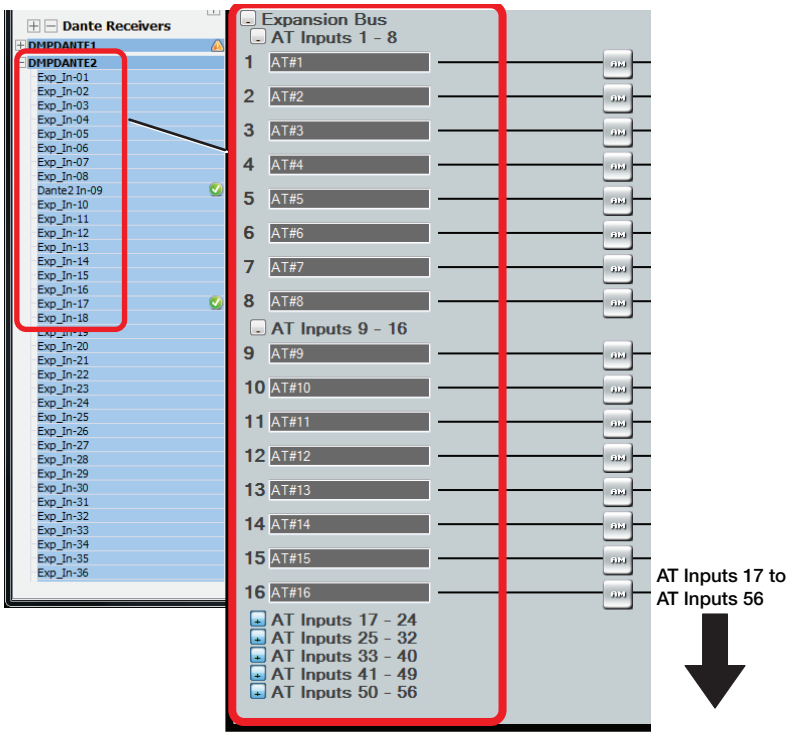


Figure 78. DSP Configurator Expansion Inputs (Dante Receiver EXP_In)

In figure 78, expansion inputs that accept signals on the AT bus from the Dante receiver channels are outlined.

NOTE: Figure 78 shows only the first 16 of 56 total expansion inputs. Routing of inputs not shown is the same.

Signal Routing with Dante

Channels either transmitted to the network, or received from the network are routed using the main Dante network view (see figure 79).

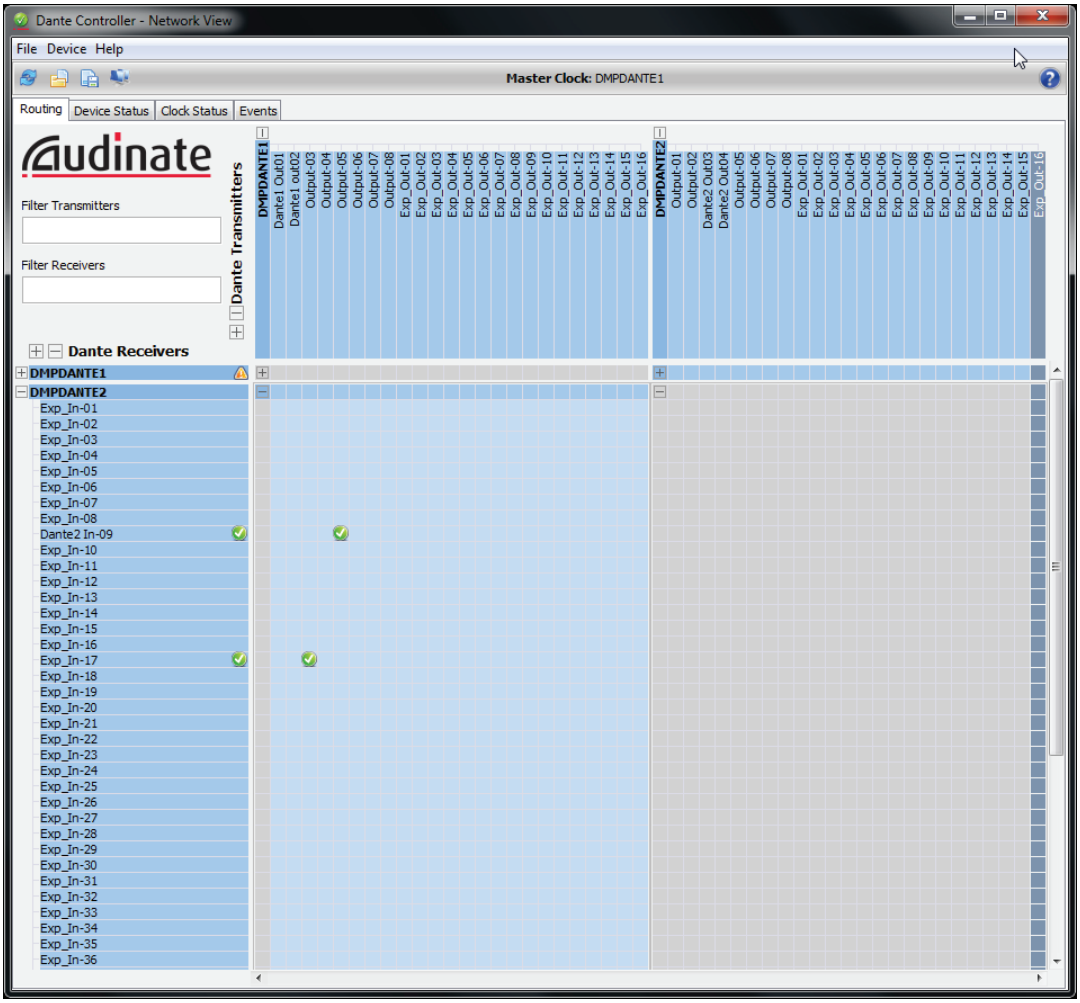


Figure 79. Dante Routing View

There are several icons used to communicate the status of both the transmitter and receiver channels and the connections made.

Icon	Description
	Indicates an active connection
	Indicates waiting for a connection to be made
	Indicates a broken connection

The transmitters along the top, and receivers down the side form a matrix on the screen whose intersection is the connection between the two. Click once at this intersection to make the connection. Click again to break the connection.

See the Dante Controller Manual at **www.audinate.com** for additional information.

Using the Extron Expansion Bus with DMP 128 AT Devices

A DMP 128P AT model can be connected to a non-AT model by the Expansion (EXP) ports with the following exceptions:

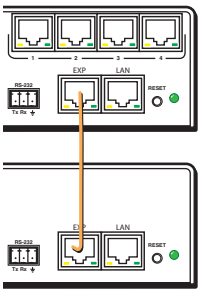
- Two AT models cannot be connected together using their EXP ports.
- AT models are always the primary unit. The non-AT model must be set as the secondary unit.
- The sixteen EXP outputs from the non-AT secondary unit connect to the AT primary unit at inputs 41 through 56.
- The secondary DMP 128 or DMP 128 C EXP inputs are routed as they normally would be.

DMP 128 AT devices are set as the Primary unit by default. This setting cannot be changed because the expansion bus primary and secondary selections in Device Settings are not available on AT models (see **Dante Network Audio Setup** on page 113). When an AT device is connected to a non-AT device by the Expansion port, the non-AT device must be set as the secondary unit.

To configure Expansion port connectivity using a DMP 128 AT:

1. Power on both units. Open DSP Configurator and connect **`Live`** to the **non-AT** model DMP 128.
2. From the menu, select **`Tools > Expansion Bus > Secondary Unit`**.
3. Connect the DMP 128P AT EXP port to the non-AT model DMP 128 EXP port using the included shielded one foot CAT 6 or similar CAT 6 cable. When connectivity is established, the front panel EXP LED on the primary unit (AT) lights solid and the EXP LED on the secondary unit blinks.

   In DSP Configurator, EXP bus channels appear as EXP Inputs 41 through 56 on AT models, and 1 through 16 on non-AT models.



NOTE: When two non-AT models are connected, channels received on the Expansion bus appear as Expansion Inputs 1 through 16.

Configuration of the EXP channels can be done at this time. You can also add additional AT models to the digital audio network. Always make the Expansion port connection between the two units before adding AT units to the digital audio network (see **Dante Network Audio Setup** on page 113).

NOTE: The Dante Controller software is required for configuration of the AT expansion bus (see **Dante Controller Software Installation** on page 113).

When connected to a non-AT device, all 56 receive channels of the DMP 128 AT still appear on the Dante digital audio network. However, Expansion inputs 41 through 56 are connected to the secondary (non-AT) device by the EXP bus, and are not operational on the Dante audio network.

Viewing AT Channels with AT Meters

The AT meters allow the user to see the signal levels of any attached Dante device. See **View** on page 22 for information on accessing the meters.

A connected Dante device is selected and all active transmit and receive channel real-time signal levels for that device are monitored.

Telephone Interface

This section describes the telephone interface and control, including:

- **Telephone Configuration**
- **Telephone Dialer**
- **Third Party Control of the Phone Dialer**

The DMP 128 provides a telephone interface with separate input and output signal processing paths. The telephone input (Rx) is identical to the other input processing paths except the AEC block is not available (see **Telephone Rx (DMP 128 C P and DMP 128 C P AT only)** on page 58). The telephone output (Tx) is tied to Output 8 and has separate but identical processing. This allows the same signal on the telephone output to be monitored by output 8.

NOTES:

- You must be in Live mode to access the phone dialer application.
- The country code must be entered before connecting the DMP 128 to a phone system.

Telephone Configuration

The phone settings dialog provides selection of the country code, updating and importing new country codes, and setting the dialer to generate a ring tone on the telephone Rx channel.

NOTE: Extron recommends the included building blocks for the phone Tx and Rx channels be inserted as the starting point for phone input and output processing (see **Signal Path Building Blocks** on page 105).

To generate a ring tone:

Select the **Allow device to generate ring tone.** checkbox to enable (checked) the ringtone. Select again to disable (unchecked) the ring tone.

To set or change the country code:

1. From the DSP Configurator toolbar main menu, select **Tools>Device Settings**, then select the **Phone Settings** tab.
2. If the correct country code is not listed after `Selected Country:`, click **Change Phone's Country Code**, then select the correct country code for your region from the dropdown list.
3. Click **Apply**, then close the dialog box.

Telephone Dialer

A telephone dialer is provided to aid setup and troubleshooting.

To use the telephone dialer:

From the main DSP Configurator screen toolbar, select **`Tools>Phone Dialer`** (see figure 80).

The phone dialer dialog box opens (see figure 81, below).

The phone dialer uses the connected telephone lines and provides the same features as a normal dialing interface to assist the installer with setup and testing.

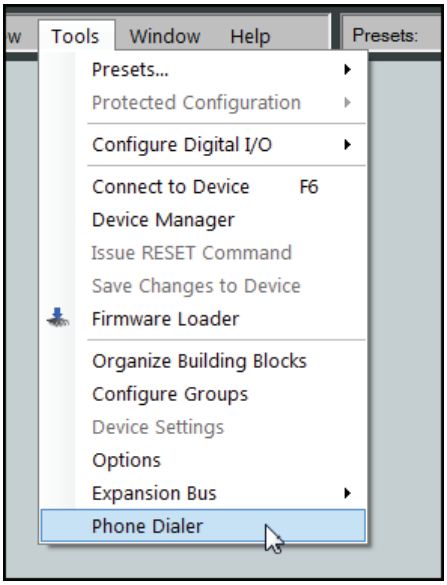


Figure 80. Select Phone Dialer

To dial a number:

- Enter a phone number using the keyboard or the number keypad, then place the phone off hook by clicking the off/on hook icon, or
- Place the phone off hook by clicking the off/on hook icon, then enter the phone number using the number keypad.

The dialer dialog has the following features:

- **Caller ID readout field** — displays the caller ID of an incoming call.
- **Phone Number readout field** — displays the last outgoing number dialed.
- **Hook Off/On Button** — toggles the phone between "Off" the hook (green-connected) or "On" (red-disconnected or hang up).
- **Number keypad** — number can be dialed before or after engaging "Hook Off".
- **Flash Button** — a single press enables call waiting or conference dialing.
- **Clear** — clears current phone number.
- **Line Status Readouts**
  - **Line Voltage** — Measures the line voltage of a connected phone line.
  - **Loop Current** — Measures the closed loop current of a connected phone line.
- **Enable Pulse Dialing** — when checked, switches the dialing method from tone to pulse.

Figure 81. Phone Dialer

Third Party Control of the Phone Dialer

See the SIS section for commands to control and query the phone dialer (see **Telephone commands** on page 146).

When sending a dial command or off hook command using SIS commands, you must first initialize the phone.

To prevent excessive levels from the dial tone, or on hook and off hook sounds, fine control of the phone receive audio is recommended during dialing. Examples of recommended phone control sequences follow:

Off hook:

1. Send receive mute SIS command to DMP.
2. Delay 100 ms.
3. Send initialize phone SIS command to DMP.
4. Send off hook SIS command to DMP.
5. Delay 100 ms.
6. Send receive gain -12 dB SIS command to DMP.

On hook:

1. Send receive mute SIS command to DMP.
2. Delay 100 ms.
3. Send on hook SIS command to DMP.
4. Delay 500 ms.
5. Send receive gain 0 dB SIS command to DMP.

Dial:

1. Send receive mute SIS command to DMP.
2. Delay 100 ms.
3. Send initialize phone SIS command to DMP.
4. Send off hook SIS command to DMP.
5. Send receive gain -12 dB SIS command to DMP.
6. Delay 200 ms.
7. Send dial number string SIS command to DMP.
8. Send receive gain 0 dB SIS command to DMP.

SIS Programming and Control

This section describes SIS programming and control of the DMP 128, including:

- **Connection Options**
- **Host-to-device Communications**
- **Command and Response Tables**

Connection Options

The DMP 128 Digital Matrix Processor can be remotely connected via a host computer or other device (such as a control system) attached to the rear panel RS-232 port or LAN port, or the front panel USB Config port.

The DMP 128 can be set up and controlled using SIS commands, embedded Web pages, or DSP Configurator software. See **Installation** on page 5 for pin assignments and details on the configuration and control port connections. For information on DSP Configurator, see **DMP Software** on page 12 and for the embedded Web pages, see **HTML Operation** on page 155.

SIS commands can be executed using the Extron Electronics DataViewer program, found on the Software Products DVD included with the product.

DMP 128 RS-232 protocol:

- 38400 baud
- 8 data bits
- no parity
- no flow control
- 1 stop bit

NOTE: The rear panel configuration port requires 38400 baud communication. This is a higher speed than many other Extron products use. The DMP 128 control software automatically sets the connection for the appropriate speed. When using DataViewer or similar application, make sure the PC or control system connected to the port is set for 38400 baud.

See **RS-232 Port** on page 130, for additional details on connecting the RS-232 port.

USB port details:

The Extron USB driver must be installed before use (see **Installing the USB Driver** on page 15).

LAN port defaults:

DMP 128 IP address:	192.168.254.254
Gateway IP address:	0.0.0.0
Subnet mask:	255.255.0.0
DHCP:	off

RS-232 Port

The DMP 128 has a serial port that can be connected to a host device such as a computer running the HyperTerminal utility, or the DataViewer utility. The port makes serial control of the switcher possible. Use the protocol information listed above to make the connection (see **Host-to-device Communications** on page 132).

USB Port (Front Panel)

The DMP 128 has a front panel USB port that can be connected to a host device such as a computer running the HyperTerminal utility, or the DataViewer utility. The port makes serial control of the switcher possible. Once the connection is established, SIS programming can begin (see **Host-to-device Communications** on page 132)

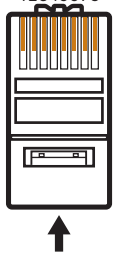
Ethernet (LAN) Port

The rear panel LAN connector on the device can be connected to an Ethernet LAN or WAN. Communication between the device and the controlling device is via Telnet (a TCP socket using port 23). The Telnet port can be changed, if necessary, via SIS. This connection makes SIS control of the device possible using a computer connected to the same LAN or WAN. The SIS commands and behavior of the product are identical to the commands and behavior the product exhibits when communicating by serial port or USB.

Ethernet Connection

The Ethernet cable can be terminated as a straight-through cable or a crossover cable and must be properly terminated for your application (see figure 82).

- **Crossover cable** — Direct connection between the computer and the DMP 128.
- **Patch (straight) cable** — Connection of the DMP 128 to an Ethernet LAN.



Crossover Cable
(for direct connection to a PC)

End 1 Pin	End 1 Wire Color	End 2 Pin	End 2 Wire Color
1	white-orange	1	white-green
2	orange	2	green
3	white-green	3	white-orange
4	blue	4	blue
5	white-blue	5	white-blue
6	green	6	orange
7	white-brown	7	white-brown
8	brown	8	brown

T568A **T568B**

A cable that is wired as TIA/EIA T568A at one end and T568B at the other (Tx and Rx pairs reversed) is a "crossover" cable.

Straight-through Cable
(for connection to a switch, hub, or router)

End 1 Pin	End 1 Wire Color	End 2 Pin	End 2 Wire Color
1	white-orange	1	white-orange
2	orange	2	orange
3	white-green	3	white-green
4	blue	4	blue
5	white-blue	5	white-blue
6	green	6	green
7	white-brown	7	white-brown
8	brown	8	brown

T568B **T568B**

A cable wired the same at both ends is called a "straight-through" cable because no pin/pair assignments are swapped.

Figure 82. RJ-45 Ethernet Connector Pin Assignments

To Establish a Network Connection to the DMP 128:

1. Open a TCP socket to port 23 using the mixer IP address.

   NOTE: If the local system administrators have not changed the value, the factory-specified default, 192.168.254.254, is the correct value for this field.

2. The DMP 128 responds with a copyright message including the date, the name of the product, firmware version, part number, and the current date/time.

   a. If the DMP 128 is not password-protected, the device is ready to accept SIS commands immediately after it sends the copyright message.
   b. If the DMP 128 is password-protected, a `password` prompt appears below the copyright message. Proceed to step 3.

3. If the device is password protected, enter the appropriate administrator or user password.

   a. If the password is accepted, the device responds with `Login User` or `Login Administrator`.
   b. If the password is not accepted, the `Password` prompt reappears.

Connection Timeouts

The Ethernet link times out after a designated period of time of no communications. By default, this timeout value is set to 5 minutes but the value can be changed (see **Ethernet data port** on page 137).

NOTE: Extron recommends leaving the default timeout at 5 minutes and periodically issuing the `Query` (Q) command to keep the connection active. If there are long idle periods, disconnect the socket and reopen the connection when another command must be sent.

Verbose Mode

Telnet connections can be used to monitor for changes that occur, such as SIS commands from other Telnet sockets or a serial port. For a Telnet session to receive change notices, the Telnet session must be in verbose mode 1 or 3 (see the **Verbose mode** SIS command on page 136). In verbose mode 1 or 3, the Telnet socket reports changes in messages that resemble SIS command responses.

Host-to-device Communications

The commands listed in the following tables perform the same functions, but are encoded differently to accommodate the requirements of each port (Telnet or browser).

DMP 128-initiated Messages

The DMP 128 initiates messages under specific conditions. No response is required from the host. The DMP 128-initiated messages are listed here (underlined).

The switcher initiates the `Copyright` message when it is first powered on or when a connection via Internet protocol (IP) is established.

`© Copyright 2Ø13, Extron Electronics, DMP 128 {`*`string`*`}, V`*`n.nn`*`, 6Ø-`*`nnnn-nn`*

`Day, DD MMM YYYY HH:MM:SS`

- **V***n.nn* is the firmware version number.
- {*string*} is the full model number of the DMP 128.
- **6Ø**-*nnnn-nn* is the DMP 128 part number.

The DMP 128 sends the boot and copyright messages under the following circumstances:

- If the DMP 128 is off and an RS-232 connection is already set up (the PC is cabled to the DMP 128 and a serial communication program such as HyperTerminal is open), the connected unit sends these messages via RS-232 when first powered on.
- If the DMP 128 is on, it sends the boot and copyright messages when a Telnet connection to the DMP 128 is first opened. The day of the week, date, and time are shown when the DMP 128 is connected via Telnet, but not via RS-232. If using a Telnet connection, the copyright message, date, and time may be followed by a password prompt.

Password Information

The "↵`Password:`" prompt requires a password (administrator level or user level) followed by a carriage return. The prompt is repeated if the correct password is not entered.

If the correct password is entered, the unit responds with "↵`Login Administrator`↵" or "↵`Login User`↵", depending on the password entered. If passwords are the same for both administrator and user, the unit will default to administrator privileges.

Using the Command and Response Tables

SIS commands consist of a string (one or more characters per command field). No special characters are required to begin or end a command sequence. When the DMP 128 determines a command is valid, it executes the command and sends a response to the host device. All responses end with a carriage return and a line feed (CR/LF = ↵), signaling the end of the response character string.

When programming, certain characters are more conveniently represented by their hexadecimal rather than ASCII values. The table below shows the hexadecimal equivalent of each ASCII character:

ASCII to HEX Conversion Table																Esc 1B		CR ØD		LF ØA	
Space	2Ø	!	21	"	22	#	23	$	24	%	25	&	26	'	27						
(	28	)	29	*	2A	+	2B	,	2C	-	2D	.	2E	/	2F						
Ø	3Ø	1	31	2	32	3	33	4	34	5	35	6	36	7	37						
8	38	9	39	:	3A	;	3B	<	3C	=	3D	>	3E	?	3F						
@	4Ø	A	41	B	42	C	43	D	44	E	45	F	46	G	47						
H	48	I	49	J	4A	K	4B	L	4C	M	4D	N	4E	O	4F						
P	5Ø	Q	51	R	52	S	53	T	54	U	55	V	56	W	57						
X	58	Y	59	Z	5A	[	5B	\	5C	]	5D	^	5E	_	5F						
`	6Ø	a	61	b	62	c	63	d	64	e	65	f	66	g	67						
h	68	i	69	j	6A	k	6B	l	6C	m	6D	n	6E	o	6F						
p	7Ø	q	71	r	72	s	73	t	74	u	75	v	76	w	77						
x	78	y	79	z	7A	{	7B	|	7C	}	7D	~	7E	DEL	7F						

Figure 83. ASCII to Hex Conversion Table

The Command and Response tables list valid ASCII (for Telnet or RS-232) command codes, the corresponding URL (uniform resource locator) encoded (for Web browsers) command codes, the DMP 128 responses to the host, and a description of the command function or the results of executing the command.

Symbol definitions

↵ = Carriage return/line feed

← = Carriage return (no line feed)

• = Space character

| = Pipe (can be used interchangeably with the ← character)

* = Asterisk character (which is a command character, not a variable)

Esc = Escape key

W = can be used interchangeably with the Esc character

NOTE: For Web encoding only: data is directed to the specified port and must be encoded (URL encoding) if it is non-alphanumeric. Change any non-alphanumeric character (%, +, ←) within the data section into the corresponding hexadecimal equivalent, %xx, where xx represents the two-character hex byte. For example, a space (hex: 2Ø) would be encoded as %2Ø and a plus sign (hex: 2B) would be encoded as %2B.

Error Responses

When the DMP 128 is unable to execute the command, it returns an error response to the host. The error response codes and their descriptions are as follows:

E12 - Invalid port number

E11 - Invalid preset

E13 - Invalid parameter (number is out of range)

E14 - Not valid for this configuration

E17 - System timed out

E22 - Busy

E24 - Privilege violation

E25 - Device is not present

E26 - Maximum connections exceeded

E27 - Invalid event number

E28 - Bad filename or file not found

Simple Control Port Commands - Telnet and Web-browser Accessible

Upper and lower case text can be used interchangeably except where noted. Port 23 is default for Telnet. Port 80 is default for web browsers. They both can be mapped to different ports.

There are minor differences when implementing these commands via Telnet or via URL encoding using a web browser. All commands listed will work using either connection method but due to some limitations of web browsers, the encapsulation characters must be modified to be certain the web browser will properly handle them. All examples are shown in a proper implementation of a Telnet or web browser session.

NOTE: When using web browsers, some non-alpha numeric characters must be represented as their hex equivalent such as %xx where xx equal the two character representation of the hex byte that needs to be sent (a comma ',' would be represented as **%2C**). Characters such as '%' (percent), '+' (plus) and ' ' (space) should also be encoded in Hex.

Telnet	**Web Browser**
Escape (Hex 1B)	W [must **not** be encoded]
Carriage Return (Hex **ØD**)	Pipe Character (**|**) [must **not** be encoded]

When describing the use of SIS commands via a web browser, the [URL] reference is used to shorten the examples. [URL] would be the full URL of the control interface and web page reference including all path information (**http://192.168.254.254/mypage.HTML**).

To send commands using a Web browser, prefix them with the full URL followed by ?cmd= (**http://192.168.254.254/mypage.html?cmd=WSF**>).

Although the DMP 128 uses the same structure for SIS commands, there are two variations. One is the global command structure noted above and documented in the **Command and Response Tables** that immediately follows.

The second set of tables, "DSP SIS commands" uses the command structure outline beginning with **DSP SIS Commands** on page 141.

Generally the basic SIS commands will be used for global configuration such as setting IP addresses, date/time, while the DSP SIS commands address audio processing functions.

Command and Response Tables

Basic SIS Commands

Command	ASCII command (host to device)	URL Encoded (web)	Response (device to host)
Information requests			
Firmware Version	Q	Q	X11↵
Firmware and build version	*Q	*Q	X11↵
Kernel firmware and build	**Q	**Q	X11↵
Verbose version info	ØQ	ØQ	Sum of 2Q-3Q-4Q↵
Firmware version	1Q	1Q	X11↵
Bootstrap Version	2Q	2Q	X11↵
Factory Firmware Version	3Q	3Q	X11 plus web ver.-desc-UL date/time↵
Updated firmware version	4Q	4Q	X11 plus web ver.-desc-UL date/time↵
NOTE: An asterisk (*) after the version number indicates the currently running version. Question marks (?.??) indicate that only factory firmware is loaded. A caret (^) indicates the firmware version that should be running, but a Mode 1 reset was executed and the default factory firmware is running. An exclamation point (!) indicates corrupted firmware.			
Query part number	N	N	DMP 128: 6Ø-1211-Ø1 DMP 128 AT: 6Ø-1211-1Ø DMP 128 C: 6Ø-1178-Ø1 DMP 128 C AT: 6Ø-1178-1Ø DMP 128 C P: 6Ø-1179-Ø1 DMP 128 C P AT: 6Ø-1179-1Ø
Query model name	I	I	VØØxØØ•A12xØ8↵
Query model name	1I	1I	DMP•128•↵ DMP•128•AT↵ DMP•128•C↵ DMP•128•C•AT↵ DMP•128•C•P↵ DMP•128•C•P•AT↵
Query model description	2I	2I	Digital•Matrix•Processor↵
Query system memory usage	3I	3I	#Bytes used out of #KBytes↵
Query user-memory usage	4I	4I	#Bytes used out of #KBytes↵

NOTE: X11 = Version number — Firmware version number to second decimal place (**x.xx**) Version and Build number adds four digits (**x.xx.xxxx**) to the Version number

2I response changes depending on the model number queried.

Command and Response Table for Basic SIS Commands (continued)

Command	ASCII command (host to device)	Response (device to host)	Additional description
IP Setup Commands			
Set unit name	Esc X12 CN ←	Ipn•X12 ↵	
View unit name	Esc CN ←	X12 ↵	
Set name to factory default	Esc •CN ←	Ipn•X49 ↵	
Set time and date	Esc X13 CT ←	Ipt•X13 ↵	
View time and date	Esc CT ←	X13 ↵	
Set GMT offset	Esc X3 CZ ←	Ipz X3 ↵	
View GMT offset	Esc CZ ←	X3 ↵	
Set Daylight Savings Time	Esc X34 CX ←	Ipx X34 ↵	
Read Daylight Savings Time	Esc CX ←	X34 ↵	
Set IP address	Esc X14 CI ←	Ipi X14 ↵	
Read IP address	Esc CI ←	X14 ↵	
Read hardware address (MAC)	Esc CH ←	X18 ↵	
Set subnet mask	Esc X19 CS ←	Ips X19 ↵	
Read subnet mask	Esc CS ←	X19 ↵	
Set gateway IP address	Esc X14 CG ←	Ipg X14 ↵	
View gateway IP address	Esc CG ←	X14 ↵	
Set DHCP on	Esc 1DH ←	Idh1 ↵	
Set DHCP off	Esc ØDH ←	IdhØ ↵	
NOTE: Changing DHCP from On to Off resets the IP address to the factory default (192.168.254.254).			
View DHCP status	Esc DH ←	X5 ↵	
Set verbose mode	Esc X22 CV ←	Vrb X22 ↵	
View verbose mode	Esc CV ←	X22 ↵	
Get connection listing	Esc CC ←	[number of connections] ↵	

NOTES:

X3 = Greenwich Mean Time offset	GMT offset value (–12:ØØ to 14:ØØ) representing hours and minutes (HH:MM) local time is offset from GMT time
X5 = On/Off status	Ø=off/disable 1=on/enable
X12 = Unit name	Alpha-numeric up to 24 characters. No special characters except hyphen (-) No upper/lower case distinction, no blanks or spaces, first character must be alpha, last character cannot be hyphen.
X13 = Local date/time	**Set:** MM/DD/YY-HH:MM:SS **Read:** day of week, date, month, year HH:MM:SS (for instance; Fri, 21 Jun 2ØØ2 1Ø:54:ØØ)
X14 = IP Address	default 192.168.254.254
X18 = Hardware MAC address	ØØ-Ø5-A6-xx-xx-xx
X19 = Subnet mask	Default 255.255.Ø.Ø
X22 = Verbose/Response mode	Ø=clear (default for IP), 1=verbose (default for serial and USB), 2=tagged responses, 3=verbose + tagged responses
X34 = Daylight Saving time	Ø=off/ignore; 1= USA (begins second Sunday in March/ends first Sunday in November); 2= Europe (begins last Sunday in March/ends last Sunday in October); 3= Brazil (begins third Sunday in October/ends third Saturday in March).
X49 = Alpha-numeric unit name	combination of unit name and last three pairs of MAC address

Command and Response Table for Basic SIS Commands (continued)

Command	ASCII command (host to device)	Response (device to host)	Additional description
Password and Security Settings			
Set administrator password	Esc X33 CA ←	Ipa•X41 ↵	
View administrator password	Esc CA ←	X41 ↵	
Reset (clear) administrator password	Esc •CA ←	Ipa• ↵	
Set user password	Esc X33 CU ←	Ipu•X41 ↵	
View user password	Esc CU ←	X41 ↵	
Reset (clear) user password	Esc •CU ←	Ipu• ↵	
Query session security level	Esc CK ←	X52 ↵	
Ethernet data port			
Set current port timeout	Esc Ø* X69 TC ←	PtiØ* X69 ↵	
View current port timeout	Esc ØTC ←	X69 ↵	
Set global IP port timeout	Esc 1* X69 TC ←	Pti1* X69 ↵	
View global IP port timeout	Esc 1TC ←	X69 ↵	
File Commands			
Erase user-supplied web page file	Esc *filename* EF ←	Del•*filename* ↵	
Erase current directory	Esc /EF ←	Ddl ↵	Also deletes files inside directory
Erase current directory and sub-directories	Esc //EF ←	Ddl ↵ *filename* x•*date/time*•*length*	
List files from current directory	Esc DF ←	*filename* x•*date/time*•*length* ↵ *filename* x•*date/time*•*length* ↵ *filename* x•*date/time*•*length* ↵ *... space_remaining*•*Bytes Left* ↵↵	
List files from current directory and below	Esc LF ←	*filename* x•*date/time*•*length* ↵ *filename* x•*date/time*•*length* ↵ *filename* x•*date/time*•*length* ↵ *... space_remaining*•*Bytes Left* ↵↵	

NOTE: LF has same response from unit as DF command, except path / directory will precede filenames for files from directories below current directory.

NOTES:
X33 = 12 alpha-numeric characters
X41 = alpha-numeric password — If a password exists, returns four **** to mask password
X52 = Security level of connection — Ø=anonymous, 11=user, 12=administrator
X69 = IP connection timeout — 1-65ØØØ steps, 1 step=1Ø seconds, default=3Ø (3ØØ seconds)

Command and Response Table for Basic SIS Commands (continued)

Command	ASCII command (host to device)	Response (device to host)	Additional description
Serial Port			
Send Data String	Esc 1*X17*X20*X21 RS X2	response ↵	
Configure parameters	Esc 1*X25,X26,X27,X28 CP ←	Cpn1•Ccp X25,X26,X27,X28 ↵	
View serial port parameters	Esc 1 CP ←	X25,X26,X27,X28 ↵	
Configure rcv timeout	Esc 1*X17*X20*X23*X21 CE ←	Cpn1•Cce X17, X20,X23,X21 ↵	
View receive timeout	Esc 1 CE ←	X17,X20,X23,X21 ↵	
Digital I/O Port			
View I/O mode	X1 [	X40 ↵	
View I/O state	X1]	X43 ↵	

NOTES:

X1 = Port Number	Ø1 to 99 up to maximum digital ports
X2 = Command data section	For web encoding only: Data will be directed to specified port and must be encoded if non-alpha numeric. Since data can include either command terminator, they must be encoded as follows when used within the data section: Space (Hex: 2Ø) is encoded as %2Ø and Plus sign (Hex: 2B) is encoded as %2B.
X17 = Command string wait time	Ø-32767 in tens of milliseconds
X20 = Character wait time	Ø-32767 in tens of milliseconds
X21 = Length of stream or delimiter	L=Byte Count (ØØ – 32767), D=decimal value for ASCII character (Ø-ØØ255)
X23 = Priority status for receiving timeouts (Default=0)	
	Ø=Send data string command parameters if they exist
	1=Configure receive timeout command parameters instead.
X25 = Baud Rate (Default=9600)	300,600,1200,1800,2400,3600,4800,7200,9600,14400,19200 38400,57600,115200
X26 = Parity (Default=N=none)	O=odd E=even N=none M=mark S=space
X27 = Data bits	7, 8 (Default=8)
X28 = Stop bits	1, 2 (Default=1)
X40 = I/O mode	Ø=Input 1=Output
X43 = I/O state	Ø=Off 1=On

Command	ASCII command (host to device)	Response (device to host)	Additional description
Event Control			
Read event buffer memory	Esc X35,X36,X37,X38E ←	X54↵	
Write event buffer memory	Esc X35,X36,X39,X38E ←	Evt X35,X36,X37,X39↵	
NOTE: Response to Write Event is padded with leading zeros for X35 and X37.			
Read string from event buffer	Esc X35,X36,X37,X44FE ←	{string}↵	
Write string to event buffer	Esc{string}*X35,X36,X37FE ←	Evt X35,X36,X37, {string}↵	
NOTE: 'F' must be capitalized to read and write strings to event buffer memory. Response to Write Event is padded with leading zeros for X35 and X37.			
Start events	Esc 1AE ←	Ego↵	
Stop events	Esc ØAE ←	Est↵	
Query # of running events	Esc AE ←	#####↵ (5 digit number)	

NOTES:
- X35 = Event number — range ØØ-99
- X36 = Event buffer — Ø=receive, 1=Unified, 2=data, 3=NVRAM
- X37 = Event buffer offset — range Ø to Max buffer size
- X38 = Event data size (case sensitive) — b=bit, B=Byte (8-bit), S=short (16-bit), L=long (32-bit)
- X39 = Event data to write
- X44 = number of Bytes to read — range 1–127
- X54 = Data element read — ASCII digits representing numeric value of data elements read from buffer (leading zeros suppressed)

Command and Response Table for Basic SIS Commands (continued)

Command	ASCII command (host to device)	Response (device to host)	Additional description
Presets, I/O Names			
Write preset name	Esc X10 , X11 NG ←	Nmg X10 , X11 ↵	
Example:	Esc 1 , Security 1 NG ←	NmgØ1 , Security 1 ↵	Name preset 1 “Security 1”.
Read preset name	Esc X10 NG ←	X11 ↵	
Example:	Esc 2NG ←	Security 2 ↵	
Recall a preset	X10 .	Rpr X10 ↵	Command character is a period
Example:	5.	Rpr5 ↵	Recall preset 5, which becomes the current configuration.
Write input name	Esc X3 , X11 NI ←	Nmi X3 , X11 ↵	
Example:	Esc 9 , Podium camNI ←	NmiØ9 , Podium cam ↵	Name input 9 “Podium cam”.
Read input name	Esc X3 NI ←	X11 ↵	
Write output name	Esc X2 , X11 NO ←	Nmo X2 , X11 ↵	
Example:	Esc 1 , Main PJ1 NO ←	NmoØ1 , Main PJ1 ↵	Name output 1 “Main PJ1”.
Read output name	Esc X2 NO ←	X11 ↵	
Resets			
Reset presets and names	Esc ZG ←	Zpg ↵	Clear all presets and their names.
Reset an individual preset	Esc X10 ZG ←	Zpg X10 ↵	Clear preset X10.
Reset a group	Esc Z X20 GRPM ←	GrpmZ X20 ↵	Delete all members from group X20, reset parameters and soft limits.
NOTE: See **Group Masters**, for more information about audio group masters.			
Reset flash	Esc ZFFF ←	Zpf ↵	Reset flash memory (erase user-supplied files).
System Reset (factory defaults)	Esc ZXXX ←	Zpx ↵	Resets all processors, level controls and mixers to default.
Reset all device settings and delete files	Esc ZY ←	Zpy ↵	
NOTE: This reset excludes IP settings such as IP address, subnet mask, gateway IP address, unit name, DHCP setting and port mapping (telnet/web/direct access) in order to preserve communication with the device. This reset is recommended after a firmware update.			
Absolute reset	Esc ZQQQ ←	Zpq ↵	Similar to **System Reset**, plus sets the IP address to 192.168.254.254 and the subnet mask to 255.255.0.0.

NOTES:

X3	= Input number	Ø1 – 12 (C P and C P AT models Ø1 – 13)
X2	= Output number	Ø1 – Ø8 (C P and C P AT models Ø1 – Ø9)
X10	= Preset #	32 maximum (Ø = current configuration)
X11	= Name	12 characters maximum
X20	= Group master group number	Ø1 – 32

DSP SIS Commands

Many digital signal processor (DSP) functions; gain, mute, group masters, and a protected configuration can be controlled using SIS commands. These commands follow the same general rules as basic SIS commands, but the variables (Xn) tend to be more complex. Also, an understanding of the audio signal flow is helpful to understanding the commands. Figure 84 shows the specific DSP functions available for SIS commands.

NOTE: Signal flow is described in greater detail in the section, **Audio level, Mix-point, Processing Blocks, and Signal Chains** on page 27.

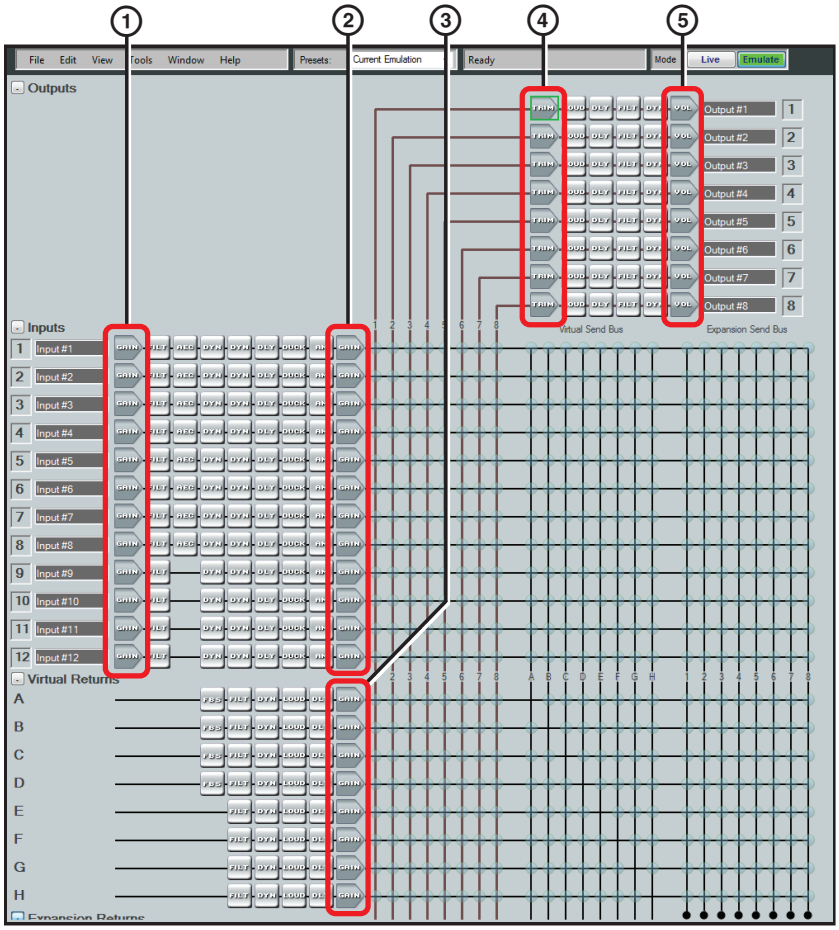


Figure 84. DSP Processors Addressable via SIS Commands

① Mic/line input gain block (including gain and mute)

② Pre-mixer gain block (including gain and mute)

③ Virtual return gain

④ Post-mixer trim (gain only)

⑤ Output Volume (including gain and mute)

Symbol Definitions

↵	=	Carriage return/line feed
←	=	Carriage return (no line feed)
•	=	Space character
|	=	Pipe (can be used interchangeably with the ← character)
*	=	Asterisk character (which is a command character, not a variable)
Esc	=	Escape key
W	=	can be used interchangeably with the Esc character

X60	=	Gain and trim control	See table 1 on page 147.
X61	=	Level value; post-mixer trim (④) mic/line gain (①) pre-mixer gain (②), virtual return gain (③), and output volume(⑤)	See table 2 beginning on page 149. -12.Ø dB to + 12.Ø dB, (1928 to 2168) in Ø.1 dB increments. –18.Ø dB to +8Ø dB, (1868 to 2848) in Ø.1 dB increments. –1ØØ.Ø dB to +12.Ø dB, (1Ø48 to 2168) in Ø.1 dB increments. **NOTE:** Output volume only, –1ØØ.Ø dB to +Ø.Ø dB (1Ø48 to 2Ø48).
X64	=	Mute status	Ø = unmute 1 = mute
X65	=	Group master group number	Ø1 – 32
X66	=	Group fader setting	dB value, in 0.1 dB increments using a 10x multiplier and negative numbers, but not decimal places. The valid range depends on the type of gain block that is assigned to the group number (X65) specified in the command: ① = –18Ø to 8ØØ (–18.0 dB to +80.0 dB) ② = –1ØØØ to 12Ø (–100.0 dB to +12.0 dB) ③ = –1ØØØ to 12Ø (-100.0 dB to + 12.0 dB) ④ = –12Ø to 12Ø (–12.0 dB to +12.0 dB) ⑤ = –1ØØØ to ØØØ (–100.0 dB to +0.0 dB) **NOTE:** Leading zeros are ignored.
X67	=	Group fader increment	dB value, in Ø.1 dB increments, to raise or lower a group fader
X68	=	Group fader soft limit	dB value, in Ø.1 dB increments. The valid range X66 must be within the range for the gain block grouped in X65.
X69	=	Group type	6= gain 12 = mute
X70	=	Personal Identification Number (PIN)	Four numeric digits, default = ØØØØ
X71	=	Protected configuration status	Ø = no protected configuration saved 1 = protected configuration saved

Special Characters

The HTML language reserves certain characters for specific functions. The device will not accept these characters as part of preset names, the device's name, passwords, or locally created file names.

The DMP 128 rejects the following characters:
{space (spaces **are** OK for names)} + ~ , @ = ‘ [ ] { } < > ’ “ ; : > \ ?

Command and Response Table for DSP SIS Commands

Command	ASCII command (host to device)	Response (device to host)	Additional description
Audio level control, and mix-point selection			
NOTES: The command format is the same, regardless of the control to be set; the acceptable adjustment range varies depending on the control or mix-point: • The mic/line input gain range is –18 dB to +8Ø dB, in Ø.1 dB increments. • The pre-mixer gain and virtual return gain range is –1ØØ dB to +12 dB, in Ø.1 dB increments. • The post-mixer trim range is –12 dB to +12 dB, in Ø.1 dB increments. • The output volume range is –1ØØ dB to Ø dB, in Ø.1 dB increments. All responses are shown with the DMP 128 in Verbose mode 2 or 3.			
Set a trim or gain	Esc G X60 * X61 AU ←	DsG X60 * X61 ↵	Set trim or mix control X60 to a value of X61 dB.
Example 1 (pre-mixer gain):	Esc G4Ø1Ø5*2Ø4ØAU ←	DsG4Ø1Ø5*2Ø4Ø↵	Set the #6 pre-mixer gain to a value of -0.8 dB.
Example 2 (mic/line input 2):	Esc G4ØØØ1*2288AU ←	DsG4ØØØ1*2288↵	Set the mic/line input 2 gain to a level of +24.0 dB.
Read a trim or gain (excluding mic/line inputs)	Esc G X60 AU ←	DsG X60 * X61 ↵	DSP trim or mix control X60 is set to a value of X61 dB.
Example 1 (post mixer gain control):	Esc G6Ø1Ø1AU ←	DsG6Ø1Ø1*21Ø3↵	Output 2, post mixer trim is set to a value of +5.5 dB.
Example 2 (mic/line input 1):	Esc G4ØØØØAU ←	DsG4ØØØØ*2598↵	Mic/line input 1 gain is set to a value of +55.0 dB.
Audio mute			
NOTES: • The post-mixer trim cannot be muted. • All responses are shown with the mixer device in Verbose mode 2 or 3.			
Audio mute	Esc M X60 *1AU ←	DsM X60 *1↵	Mute audio at X60.
Example:	Esc M6Ø1ØØ*1AU ←	DsM6Ø1ØØ*1↵	Mute output 1.
Audio unmute	Esc M X60 *ØAU ←	DsM X60 *Ø↵	Unmute audio at X60.
Read audio mute	Esc M X60 AU ←	DsM X60 * X64 ↵	X64: Ø = mute off, 1 = mute on.

NOTES:				
	X60	=	Gain and trim control	See table 1 on page 147.
	X61	=	Level value: post-mixer trim, mic/line gain, pre-mixer gain, virtual return gain, and output volume	See table 2 beginning on page 149.
	X64	=	Mute status	0 = unmute 1 = mute

Command and Response Table for DSP SIS Commands (continued)

Command	ASCII command (host to device)	Response (device to host)	Additional description
Audio group master commands			
NOTES: • See **Group Masters**, for more information about audio group masters. • A group must have assigned members for these commands to have an effect. • For X66, a positive (+) value is assumed unless a negative (-) value is specified. • If entering a X66 value outside the valid range for the group or outside the soft limits, the DMP 128 responds with an "invalid parameter" (E13) error. • X66, X67, and X68 values can be sent without leading zeroes; responses are always 5 digits.			
Set a group fader control	Esc D X65 * X66 GRPM ←	GrpmD X65 * X66 ↵	Set a group fader to a value of X66.
Example:	Esc d2*-293*GRPM ←	GrpmDØ2*-ØØ293↵	Set the group 2 fader control to -29.3 dB.
Raise a group fader control	Esc D X65 * X67 +GRPM ←	GrpmD X65 * X66 ↵	Increase the level of the X65 group fader by X67 dB.
Example:	Esc d2*3Ø+GRPM ←	GrpmDØ2*-ØØ263↵	Raise the group 2 fader 3 dB (from -29.3 dB to -26.3 dB, starting from the level set in the "Set a group fader control" example, above.
Lower a group fader control	Esc D X65 * X67 -GRPM ←	GrpmD X65 * X66 ↵	Decrease the level of the X65 group fader by X67 dB.
View the group fader control level	Esc D X65 GRPM ←	GrpmD X65 * X66 ↵	In verbose modes 1 and 2, the response is simplified to X66↵.
Mute a group mute control	Esc D X65 *1GRPM ←	GrpmD X65 *+ØØØØ1↵	Mute all blocks in group X65.
Clear (unmute) a group mute control	Esc D X65 *ØGRPM ←	GrpmD X65 *+ØØØØØ↵	Umute all blocks in group X65.
View a group mute control	Esc D X65 GRPM ←	GrpmD X65 * X64 ↵	For group masters, X64 is always expressed as a positive or negative 5-digit value.
Set soft limits	Esc L X65 * X68 upper * X68 lower GRPM ←	GrpmL X65 * X68 * X68 ↵	Set the groups soft limits to X68 and X68.
Example:	Esc L2*+6Ø*-6ØGRPM ←	GrpmLØ2*+ØØØ6Ø*-ØØØ6Ø↵	Set the upper soft limit for the group 2 fader to +6.0 dB and the lower limit to -6.0 dB.
View soft limits	Esc L X65 GRPM ←	GrpmL X65 * X68 * X68 ↵	In verbose modes 0 and 1, the response is simplified to X68*X68↵.
View group type	Esc P X65 GRPM ←	GrpmP X65 * X69 ↵	Show the group type (X69) for group X65. In verbose modes 0 and 1, the response is simplified to X69↵.
View group members	Esc O X65 GRPM ←	GrpmO X65 * X60 1 * X60 2 * ... * X60 16 ↵	X60 is the control address. In verbose modes 0 and 1, the response is simplified to X60 1 * X60 2 * ... * X60 16 ↵.

NOTES:

X60 = Gain and trim control	See table 1 on page 144
X65 = Group master group number	Ø1 - 32
X66 = Group fader level	dB value, in 0.1 dB increments, using negative numbers and eliminating decimal places using a 10x multiplier. –100.0 dB to +80.0 dB is represented by –1000 to 800. The valid range depends on the type of gain or trim block assigned to the group number (X65).
X67 = Group fader increase/decrease	dB value, in 0.1 dB increments, to raise or lower a group fader.
X68 = Group fader soft limit	dB value, in 0.1 dB increments. The valid range must be within the range for the gain block grouped in X65.
X69 = Group type	6 = gain 12 = mute

Command and Response Table for DSP SIS Commands (continued)

Command	ASCII command (host to device)	Response (device to host)	Additional description
Protected configuration			
NOTE: The DMP 128 can save and recall a Personal Identification Number (PIN)-protected configuration, which includes all processing and levels in the current coinfiguration, with the exception of the IP address). The protected configuration is useful to establish the DMP 128 in a known state, either as a troubleshooting tool or as a baseline configuration.			
Save the configuration	Esc S X70 PCFG ←	PcfgS ↵	Save the configuration to the protected memory location.
Recall the configuration	Esc RPCFG ←	PcfgR ↵	Recall the protected configuration
Change the PIN	Esc P X70^{old}* X70^{new} PCFG ←	PcfgP X70^{new} ↵	Overwrite the old PIN (X70^{old}) with the new one (X70^{new}).
Query configuration saved status	Esc QPCFG ←	X71 ↵	

NOTES: X70 = Personal Identification Number (PIN) — Four numeric digits, default = ØØØØ
X71 = Protected configuration status — Ø = no protected configuration saved; 1 = protected configuration saved

Command	ASCII command (host to device)	Response (device to host)	Additional description
Telephone commands			
NOTE: Prior to using the the modem, ensure the country code is set using DSP Configurator (see **Telephone Configuration** on page 126). To receive proper responses, place the DMP 128 in verbose mode 3.			
Initialize phone	Esc AT+FCLASS=8,PHON←	PhonOK↵	Must be sent prior to dialing or going off hook
Dial number string	Esc ATDT X2 ,PHON←	PhonOK ↵	Send DTMF string X2 to modem
Off hook	Esc AT+VLS=13,PHON←	PhonOffHook↵	Connects line
On hook	Esc AT+VLS=Ø, PHON←	PhonOnHook↵	Disconnects line
Flash	Esc ATD&,PHON←	PhonOK ↵	Flashes line
Phone state	Esc AT+VLS?,PHON←	PhonSts X1 ↵	X1 ØØ = On hook 13 = Off hook
Dial single digit	Esc AT. X3 ,PHON←	PhonOK ↵	Send single DTMF pulse X3 to modem
Receive gain 0 dB	Esc AT:UB5%2C1ØØØ%ØD,1PHON←	PhonOK ↵	Sets receive gain to 0 dB
Receive gain -12 dB	Esc AT:UB5%2C4ØØ%ØD,1PHON←	PhonOK ↵	Sets receive gain to -12 dB
Receive mute	Esc AT:UB5%2CØ%ØD,1PHON←	PhonOK ↵	Mutes the phone receiver
Telephone responses (unsolicited)			
NOTE: The following unsolicited responses alert the DMP 128 control system of an incoming call, line status changes, or an incoming Caller ID. To receive proper responses, place the DMP 128 in verbose mode 3.			
		PhonDial_Tone ↵	Line is off hook, dial tone present
		PhonRinging_Tone ↵	Far end ringing
		PhonBusy_Tone ↵	Far end busy
		PhonLine_Intr ↵	Far end has terminated the call
		PhonError_Non_Voice ↵	Phone was not initialized properly
		PhonError ↵	Unrecognized phone command
		PhonRING ↵	Incoming call
		PhonDATE=X5 ↵	From Caller ID of incoming call, X5 = Current Date
		PhonTIME=X6 ↵	From Caller ID of incoming call, X6 = Current time
		PhonNAME=X7 ↵	From Caller ID of incoming call, X7 = Name
		PhonNMBR=X2 ↵	From Caller ID of incoming call, X2 = Phone Number

NOTES:	X1 = Hook status	ØØ=on hook, 13= off hook
	X2 = Numeric string	Numeric string. Valid characters are: Ø through 9,*,#, no spaces
	X3 = Single character	Valid characters are: Ø through 9,*,#
	X5 = Date	Date in mmdd format
	X6 = Current time	Caller ID current time in hhss format
	X7 = Character String	Up to 10 characters

Table 1: Level Control

① Input Gain control	X60	Level
Mic/Line Input 1	40000	X61
Mic/Line Input 2	40001	X61
Mic/Line Input 3	40002	X61
Mic/Line Input 4	40003	X61
Mic/Line Input 5	40004	X61
Mic/Line Input 6	40005	X61
Mic/Line Input 7	40006	X61
Mic/Line Input 8	40007	X61
Mic/Line Input 9	40008	X61
Mic/Line Input 10	40009	X61
Mic/Line Input 11	4001Ø	X61
Mic/Line Input 12	40011	X61
Telephone Rx	40012	X61

② Pre-mixer gain	X60	Level
Mic/Line Input 1	40100	X61
Mic/Line Input 2	40101	X61
Mic/Line Input 3	40102	X61
Mic/Line Input 4	40103	X61
Mic/Line Input 5	40104	X61
Mic/Line Input 6	40105	X61
Mic/Line Input 7	40106	X61
Mic/Line Input 8	40107	X61
Mic/Line Input 9	40108	X61
Mic/Line Input 10	40109	X61
Mic/Line Input 11	4011Ø	X61
Mic/Line Input 12	40111	X61
Telephone Rx	40112	X61

④ Post-mixer trim	X60	Level
Output 1	60100	X61
Output 2	6Ø101	X61
Output 3	6Ø102	X61
Output 4	6Ø103	X61
Output 5	6Ø104	X61
Output 6	6Ø105	X61
Output 7	6Ø106	X61
Output 8/Telephone Tx	6Ø107	X61

⑤ Volume Out Control	X60	Level
Output 1	60000	X61
Output 2	60001	X61
Output 3	60002	X61
Output 4	60003	X61
Output 5	60004	X61
Output 6	60005	X61
Output 7	60006	X61
Output 8	60007	X61
Telephone Tx	60008	X61

③ Virtual Return Gain	X60	Level
Virtual Return A	50000	X61
Virtual Return B	50001	X61
Virtual Return C	50002	X61
Virtual Return D	50003	X61
Virtual Return E	50004	X61
Virtual Return F	50005	X61
Virtual Return G	50006	X61
Virtual Return H	50007	X61

NOTE:

- For expanded coverage of setting audio levels, including comprehensive tables of mix-point address, see **DSP SIS Commands for Mix-point Level and Mute Control** on page 170.
- Telephone Tx and Rx commands available on DMP 128 P models only.

Setting Audio Levels

The audio dB to SIS command conversion values in table 2 beginning on the next page, are the same for all signal level blocks. However, the minimum and maximum levels differ depending upon the individual level control. The following table can determine those minimum and maximum levels for the individual controls. Once the range and a value within that range is selected, table 2 is used to find the SIS value for the desired dB level.

SIS	Title	Function	Range
X61	Mic/line gain (①)	Attenuation/Gain	–18.Ø to +8Ø.Ø dB
	Pre-mixer gain (②)	Attenuation/Gain	–1ØØ.Ø to +12.Ø dB
	Virtual return gain (③)	Attenuation/Gain	–1ØØ.Ø to +12.Ø dB
	Post-mixer trim (④)	Attenuation/Gain	–12.Ø to +12.Ø dB
	Output volume (⑤)	Attenuation	–1ØØ.Ø to +Ø.Ø dB

Alternately, the following formula is used to convert a desired dB value into the proper SIS command:

(Desired dB x 10) + 2048

Examples:

SIS value for 0 dB:

- (0 x 10) + 2048 = 0 + 2048 = 2048

SIS value for +7.0 dB:

- (+7.0 x 10) + 2048 = 70 + 2048 = 2118

SIS value for +22.5 dB:

- (+22.5 x 10) + 2048 = 225 + 2048 = 2273

SIS value for -12.0 dB:

- (-12.0 x 10) + 2048= -120 + 2048 = 1928

Table 2: Audio dB to SIS Command Conversion Values

NOTES:

- The maximum range of the mic/line gain (①) is -18.Ø to +8Ø.Ø dB.
- The maximum range of the pre mixer gain (②) is -1ØØ.Ø to +12.Ø dB.
- The maximum range of the virtual return gain (③) is -1ØØ.Ø to +12.Ø dB.
- The maximum range of the post mixer trim control (④) is -12.Ø to +12.Ø dB.
- The maximum range of the output volume control (⑤) is -1ØØ.Ø to +Ø.Ø dB.

dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61
																		-1ØØØ.Ø	1Ø48
-99.9	1Ø49	-99.8	1Ø5Ø	-99.7	1Ø51	-99.6	1Ø52	-99.5	1Ø53	-99.4	1Ø54	-99.3	1Ø55	-99.2	1Ø56	-99.1	1Ø57	-99.Ø	1Ø58
-98.9	1Ø59	-98.8	1Ø6Ø	-98.7	1Ø61	-98.6	1Ø62	-98.5	1Ø63	-98.4	1Ø64	-98.3	1Ø65	-98.2	1Ø66	-98.1	1Ø67	-98.Ø	1Ø68
-97.9	1Ø69	-97.8	1Ø7Ø	-97.7	1Ø71	-97.6	1Ø72	-97.5	1Ø73	-97.4	1Ø74	-97.3	1Ø75	-97.2	1Ø76	-97.1	1Ø77	-97.Ø	1Ø78
-96.9	1Ø79	-96.8	1Ø8Ø	-96.7	1Ø81	-96.6	1Ø82	-96.5	1Ø83	-96.4	1Ø84	-96.3	1Ø85	-96.2	1Ø86	-96.1	1Ø87	-96.Ø	1Ø88
-95.9	1Ø89	-95.8	1Ø9Ø	-95.7	1Ø91	-95.6	1Ø92	-95.5	1Ø93	-95.4	1Ø94	-95.3	1Ø95	-95.2	1Ø96	-95.1	1Ø97	-95.Ø	1Ø98
-94.9	1Ø99	-94.8	11ØØ	-94.7	11Ø1	-94.6	11Ø2	-94.5	11Ø3	-94.4	11Ø4	-94.3	11Ø5	-94.2	11Ø6	-94.1	11Ø7	-94.Ø	11Ø8
-93.9	11Ø9	-93.8	111Ø	-93.7	1111	-93.6	1112	-93.5	1113	-93.4	1114	-93.3	1115	-93.2	1116	-93.1	1117	-93.Ø	1118
-92.9	1119	-92.8	112Ø	-92.7	1121	-92.6	1122	-92.5	1123	-92.4	1124	-92.3	1125	-92.2	1126	-92.1	1127	-92.Ø	1128
-91.9	1129	-91.8	113Ø	-91.7	1131	-91.6	1132	-91.5	1133	-91.4	1134	-91.3	1135	-91.2	1136	-91.1	1137	-91.Ø	1138
-9Ø.9	1139	-9Ø.8	114Ø	-9Ø.7	1141	-9Ø.6	1142	-9Ø.5	1143	-9Ø.4	1144	-9Ø.3	1145	-9Ø.2	1146	-9Ø.1	1147	-9Ø.Ø	1148
-89.9	1149	-89.8	115Ø	-89.7	1151	-89.6	1152	-89.5	1153	-89.4	1154	-89.3	1155	-89.2	1156	-89.1	1157	-89.Ø	1158
-88.9	1159	-88.8	116Ø	-88.7	1161	-88.6	1162	-88.5	1163	-88.4	1164	-88.3	1165	-88.2	1166	-88.1	1167	-88.Ø	1168
-87.9	1169	-87.8	117Ø	-87.7	1171	-87.6	1172	-87.5	1173	-87.4	1174	-87.3	1175	-87.2	1176	-87.1	1177	-87.Ø	1178
-86.9	1179	-86.8	118Ø	-86.7	1181	-86.6	1182	-86.5	1183	-86.4	1184	-86.3	1185	-86.2	1186	-86.1	1187	-86.Ø	1188
-85.9	1189	-85.8	119Ø	-85.7	1191	-85.6	1192	-85.5	1193	-85.4	1194	-85.3	1195	-85.2	1196	-85.1	1197	-85.Ø	1198
-84.9	1199	-84.8	12ØØ	-84.7	12Ø1	-84.6	12Ø2	-84.5	12Ø3	-84.4	12Ø4	-84.3	12Ø5	-84.2	12Ø6	-84.1	12Ø7	-84.Ø	12Ø8
-83.9	12Ø9	-83.8	121Ø	-83.7	1211	-83.6	1212	-83.5	1213	-83.4	1214	-83.3	1215	-83.2	1216	-83.1	1217	-83.Ø	1218
-82.9	1219	-82.8	122Ø	-82.7	1221	-82.6	1222	-82.5	1223	-82.4	1224	-82.3	1225	-82.2	1226	-82.1	1227	-82.Ø	1228
-81.9	1229	-81.8	123Ø	-81.7	1231	-81.6	1232	-81.5	1233	-81.4	1234	-81.3	1235	-81.2	1236	-81.1	1237	-81.Ø	1238
-8Ø.9	1239	-8Ø.8	124Ø	-8Ø.7	1241	-8Ø.6	1242	-8Ø.5	1243	-8Ø.4	1244	-8Ø.3	1245	-8Ø.2	1246	-8Ø.1	1247	-8Ø.Ø	1248
-79.9	1249	-79.8	125Ø	-79.7	1251	-79.6	1252	-79.5	1253	-79.4	1254	-79.3	1255	-79.2	1256	-79.1	1257	-79.Ø	1258
-78.9	1259	-78.8	126Ø	-78.7	1261	-78.6	1262	-78.5	1263	-78.4	1264	-78.3	1265	-78.2	1266	-78.1	1267	-78.Ø	1268
-77.9	1269	-77.8	127Ø	-77.7	1271	-77.6	1272	-77.5	1273	-77.4	1274	-77.3	1275	-77.2	1276	-77.1	1277	-77.Ø	1278
-76.9	1279	-76.8	128Ø	-76.7	1281	-76.6	1282	-76.5	1283	-76.4	1284	-76.3	1285	-76.2	1286	-76.1	1287	-76.Ø	1288
-75.9	1289	-75.8	129Ø	-75.7	1291	-75.6	1292	-75.5	1293	-75.4	1294	-75.3	1295	-75.2	1296	-75.1	1297	-75.Ø	1298
-74.9	1299	-74.8	13ØØ	-74.7	13Ø1	-74.6	13Ø2	-74.5	13Ø3	-74.4	13Ø4	-74.3	13Ø5	-74.2	13Ø6	-74.1	13Ø7	-74.Ø	13Ø8

dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61
−73.9	1309	−73.8	1310	−73.7	1311	−73.6	1312	−73.5	1313	−73.4	1314	−73.3	1315	−73.2	1316	−73.1	1317	−73.Ø	1318
−72.9	1319	−72.8	1320	−72.7	1321	−72.6	1322	−72.5	1323	−72.4	1324	−72.3	1325	−72.2	1326	−72.1	1327	−72.Ø	1328
−71.9	1329	−71.8	1330	−71.7	1331	−71.6	1332	−71.5	1333	−71.4	1334	−71.3	1335	−71.2	1336	−71.1	1337	−71.Ø	1338
−70.9	1339	−70.8	1340	−70.7	1341	−70.6	1342	−70.5	1343	−70.4	1344	−70.3	1345	−70.2	1346	−70.1	1347	−70.Ø	1348
−69.9	1349	−69.8	1350	−69.7	1351	−69.6	1352	−69.5	1353	−69.4	1354	−69.3	1355	−69.2	1356	−69.1	1357	−69.Ø	1358
−68.9	1359	−68.8	1360	−68.7	1361	−68.6	1362	−68.5	1363	−68.4	1364	−68.3	1365	−68.2	1366	−68.1	1367	−68.Ø	1368
−67.9	1369	−67.8	1370	−67.7	1371	−67.6	1372	−67.5	1373	−67.4	1374	−67.3	1375	−67.2	1376	−67.1	1377	−67.Ø	1378
−66.9	1379	−66.8	1380	−66.7	1381	−66.6	1382	−66.5	1383	−66.4	1384	−66.3	1385	−66.2	1386	−66.1	1387	−66.Ø	1388
−65.9	1389	−65.8	1390	−65.7	1391	−65.6	1392	−65.5	1393	−65.4	1394	−65.3	1395	−65.2	1396	−65.1	1397	−65.Ø	1398
−64.9	1399	−64.8	1400	−64.7	1401	−64.6	1402	−64.5	1403	−64.4	1404	−64.3	1405	−64.2	1406	−64.1	1407	−64.Ø	1408
−63.9	1409	−63.8	1410	−63.7	1411	−63.6	1412	−63.5	1413	−63.4	1414	−63.3	1415	−63.2	1416	−63.1	1417	−63.Ø	1418
−62.9	1419	−62.8	1420	−62.7	1421	−62.6	1422	−62.5	1423	−62.4	1424	−62.3	1425	−62.2	1426	−62.1	1427	−62.Ø	1428
−61.9	1429	−61.8	1430	−61.7	1431	−61.6	1432	−61.5	1433	−61.4	1434	−61.3	1435	−61.2	1436	−61.1	1437	−61.Ø	1438
−60.9	1439	−60.8	1440	−60.7	1441	−60.6	1442	−60.5	1443	−60.4	1444	−60.3	1445	−60.2	1446	−60.1	1447	−60.Ø	1448
−59.9	1449	−59.8	1450	−59.7	1451	−59.6	1452	−59.5	1453	−59.4	1454	−59.3	1455	−59.2	1456	−59.1	1457	−59.Ø	1458
−58.9	1459	−58.8	1460	−58.7	1461	−58.6	1462	−58.5	1463	−58.4	1464	−58.3	1465	−58.2	1466	−58.1	1467	−58.Ø	1468
−57.9	1469	−57.8	1470	−57.7	1471	−57.6	1472	−57.5	1473	−57.4	1474	−57.3	1475	−57.2	1476	−57.1	1477	−57.Ø	1478
−56.9	1479	−56.8	1480	−56.7	1481	−56.6	1482	−56.5	1483	−56.4	1484	−56.3	1485	−56.2	1486	−56.1	1487	−56.Ø	1488
−55.9	1489	−55.8	1490	−55.7	1491	−55.6	1492	−55.5	1493	−55.4	1494	−55.3	1495	−55.2	1496	−55.1	1497	−55.Ø	1498
−54.9	1499	−54.8	1500	−54.7	1501	−54.6	1502	−54.5	1503	−54.4	1504	−54.3	1505	−54.2	1506	−54.1	1507	−54.Ø	1508
−53.9	1509	−53.8	1510	−53.7	1511	−53.6	1512	−53.5	1513	−53.4	1514	−53.3	1515	−53.2	1516	−53.1	1517	−53.Ø	1518
−52.9	1519	−52.8	1520	−52.7	1521	−52.6	1522	−52.5	1523	−52.4	1524	−52.3	1525	−52.2	1526	−52.1	1527	−52.Ø	1528
−51.9	1529	−51.8	1530	−51.7	1531	−51.6	1532	−51.5	1533	−51.4	1534	−51.3	1535	−51.2	1536	−51.1	1537	−51.Ø	1538
−50.9	1539	−50.8	1540	−50.7	1541	−50.6	1542	−50.5	1543	−50.4	1544	−50.3	1545	−50.2	1546	−50.1	1547	−50.Ø	1548
−49.9	1549	−49.8	1550	−49.7	1551	−49.6	1552	−49.5	1553	−49.4	1554	−49.3	1555	−49.2	1556	−49.1	1557	−49.Ø	1558
−48.9	1559	−48.8	1560	−48.7	1561	−48.6	1562	−48.5	1563	−48.4	1564	−48.3	1565	−48.2	1566	−48.1	1567	−48.Ø	1568
−47.9	1569	−47.8	1570	−47.7	1571	−47.6	1572	−47.5	1573	−47.4	1574	−47.3	1575	−47.2	1576	−47.1	1577	−47.Ø	1578
−46.9	1579	−46.8	1580	−46.7	1581	−46.6	1582	−46.5	1583	−46.4	1584	−46.3	1585	−46.2	1586	−46.1	1587	−46.Ø	1588
−45.9	1589	−45.8	1590	−45.7	1591	−45.6	1592	−45.5	1593	−45.4	1594	−45.3	1595	−45.2	1596	−45.1	1597	−45.Ø	1598
−44.9	1599	−44.8	1600	−44.7	1601	−44.6	1602	−44.5	1603	−44.4	1604	−44.3	1605	−44.2	1606	−44.1	1607	−44.Ø	1608
−43.9	1609	−43.8	1610	−43.7	1611	−43.6	1612	−43.5	1613	−43.4	1614	−43.3	1615	−43.2	1616	−43.1	1617	−43.Ø	1618
−42.9	1619	−42.8	1620	−42.7	1621	−42.6	1622	−42.5	1623	−42.4	1624	−42.3	1625	−42.2	1626	−42.1	1627	−42.Ø	1628
−41.9	1629	−41.8	1630	−41.7	1631	−41.6	1632	−41.5	1633	−41.4	1634	−41.3	1635	−41.2	1636	−41.1	1637	−41.Ø	1638

dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61
−4Ø.9	1639	−4Ø.8	164Ø	−4Ø.7	1641	−4Ø.6	1642	−4Ø.5	1643	−4Ø.4	1644	−4Ø.3	1645	−4Ø.2	1646	−4Ø.1	1647	−4Ø.Ø	1648
−39.9	1649	−39.8	165Ø	−39.7	1651	−39.6	1652	−39.5	1653	−39.4	1654	−39.3	1655	−39.2	1656	−39.1	1657	−39.Ø	1658
−38.9	1659	−38.8	166Ø	−38.7	1661	−38.6	1662	−38.5	1663	−38.4	1664	−38.3	1665	−38.2	1666	−38.1	1667	−38.Ø	1668
−37.9	1669	−37.8	167Ø	−37.7	1671	−37.6	1672	−37.5	1673	−37.4	1674	−37.3	1675	−37.2	1676	−37.1	1677	−37.Ø	1678
−36.9	1679	−36.8	168Ø	−36.7	1681	−36.6	1682	−36.5	1683	−36.4	1684	−36.3	1685	−36.2	1686	−36.1	1687	−36.Ø	1688
−35.9	1689	−35.8	169Ø	−35.7	1691	−35.6	1692	−35.5	1693	−35.4	1694	−35.3	1695	−35.2	1696	−35.1	1697	−35.Ø	1698
−34.9	1699	−34.8	17ØØ	−34.7	17Ø1	−34.6	17Ø2	−34.5	17Ø3	−34.4	17Ø4	−34.3	17Ø5	−34.2	17Ø6	−34.1	17Ø7	−34.Ø	17Ø8
−33.9	17Ø9	−33.8	171Ø	−33.7	1711	−33.6	1712	−33.5	1713	−33.4	1714	−33.3	1715	−33.2	1716	−33.1	1717	−33.Ø	1718
−32.9	1719	−32.8	172Ø	−32.7	1721	−32.6	1722	−32.5	1723	−32.4	1724	−32.3	1725	−32.2	1726	−32.1	1727	−32.Ø	1728
−31.9	1729	−31.8	173Ø	−31.7	1731	−31.6	1732	−31.5	1733	−31.4	1734	−31.3	1735	−31.2	1736	−31.1	1737	−31.Ø	1738
−3Ø.9	1739	−3Ø.8	174Ø	−3Ø.7	1741	−3Ø.6	1742	−3Ø.5	1743	−3Ø.4	1744	−3Ø.3	1745	−3Ø.2	1746	−3Ø.1	1747	−3Ø.Ø	1748
−29.9	1749	−29.8	175Ø	−29.7	1751	−29.6	1752	−29.5	1753	−29.4	1754	−29.3	1755	−29.2	1756	−29.1	1757	−29.Ø	1758
−28.9	1759	−28.8	176Ø	−28.7	1761	−28.6	1762	−28.5	1763	−28.4	1764	−28.3	1765	−28.2	1766	−28.1	1767	−28.Ø	1768
−27.9	1769	−27.8	177Ø	−27.7	1771	−27.6	1772	−27.5	1773	−27.4	1774	−27.3	1775	−27.2	1776	−27.1	1777	−27.Ø	1778
−26.9	1779	−26.8	178Ø	−26.7	1781	−26.6	1782	−26.5	1783	−26.4	1784	−26.3	1785	−26.2	1786	−26.1	1787	−26.Ø	1788
−25.9	1789	−25.8	179Ø	−25.7	1791	−25.6	1792	−25.5	1793	−25.4	1794	−25.3	1795	−25.2	1796	−25.1	1797	−25.Ø	1798
−24.9	1799	−24.8	18ØØ	−24.7	18Ø1	−24.6	18Ø2	−24.5	18Ø3	−24.4	18Ø4	−24.3	18Ø5	−24.2	18Ø6	−24.1	18Ø7	−24.Ø	18Ø8
−23.9	18Ø9	−23.8	181Ø	−23.7	1811	−23.6	1812	−23.5	1813	−23.4	1814	−23.3	1815	−23.2	1816	−23.1	1817	−23.Ø	1818
−22.9	1819	−22.8	182Ø	−22.7	1821	−22.6	1822	−22.5	1823	−22.4	1824	−22.3	1825	−22.2	1826	−22.1	1827	−22.Ø	1828
−21.9	1829	−21.8	183Ø	−21.7	1831	−21.6	1832	−21.5	1833	−21.4	1834	−21.3	1835	−21.2	1836	−21.1	1837	−21.Ø	1838
−2Ø.9	1839	−2Ø.8	184Ø	−2Ø.7	1841	−2Ø.6	1842	−2Ø.5	1843	−2Ø.4	1844	−2Ø.3	1845	−2Ø.2	1846	−2Ø.1	1847	−2Ø.Ø	1848
−19.9	1849	−19.8	185Ø	−19.7	1851	−19.6	1852	−19.5	1853	−19.4	1854	−19.3	1855	−19.2	1856	−19.1	1857	−19.Ø	1858
−18.9	1859	−18.8	186Ø	−18.7	1861	−18.6	1862	−18.5	1863	−18.4	1864	−18.3	1865	−18.2	1866	−18.1	1867	−18.Ø	1868
−17.9	1869	−17.8	187Ø	−17.7	1871	−17.6	1872	−17.5	1873	−17.4	1874	−17.3	1875	−17.2	1876	−17.1	1877	−17.Ø	1878
−16.9	1879	−16.8	188Ø	−16.7	1881	−16.6	1882	−16.5	1883	−16.4	1884	−16.3	1885	−16.2	1886	−16.1	1887	−16.Ø	1888
−15.9	1889	−15.8	189Ø	−15.7	1891	−15.6	1892	−15.5	1893	−15.4	1894	−15.3	1895	−15.2	1896	−15.1	1897	−15.Ø	1898
−14.9	1899	−14.8	19ØØ	−14.7	19Ø1	−14.6	19Ø2	−14.5	19Ø3	−14.4	19Ø4	−14.3	19Ø5	−14.2	19Ø6	−14.1	19Ø7	−14.Ø	19Ø8
−13.9	19Ø9	−13.8	191Ø	−13.7	1911	−13.6	1912	−13.5	1913	−13.4	1914	−13.3	1915	−13.2	1916	−13.1	1917	−13.Ø	1918
−12.9	1919	−12.8	192Ø	−12.7	1921	−12.6	1922	−12.5	1923	−12.4	1924	−12.3	1925	−12.2	1926	−12.1	1927	−12.Ø	1928
−11.9	1929	−11.8	193Ø	−11.7	1931	−11.6	1932	−11.5	1933	−11.4	1934	−11.3	1935	−11.2	1936	−11.1	1937	−11.Ø	1938
−1Ø.9	1939	−1Ø.8	194Ø	−1Ø.7	1941	−1Ø.6	1942	−1Ø.5	1943	−1Ø.4	1944	−1Ø.3	1945	−1Ø.2	1946	−1Ø.1	1947	−1Ø.Ø	1948
−9.9	1949	−9.8	195Ø	−9.7	1951	−9.6	1952	−9.5	1953	−9.4	1954	−9.3	1955	−9.2	1956	−9.1	1957	−9.Ø	1958

dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61
−8.9	1959	−8.8	196Ø	−8.7	1961	−8.6	1962	−8.5	1963	−8.4	1964	−8.3	1965	−8.2	1966	−8.1	1967	−8.Ø	1968
−7.9	1969	−7.8	197Ø	−7.7	1971	−7.6	1972	−7.5	1973	−7.4	1974	−7.3	1975	−7.2	1976	−7.1	1977	−7.Ø	1978
−6.9	1979	−6.8	198Ø	−6.7	1981	−6.6	1982	−6.5	1983	−6.4	1984	−6.3	1985	−6.2	1986	−6.1	1987	−6.Ø	1988
−5.9	1989	−5.8	199Ø	−5.7	1991	−5.6	1992	−5.5	1993	−5.4	1994	−5.3	1995	−5.2	1996	−5.1	1997	−5.Ø	1998
−4.9	1999	−4.8	200Ø	−4.7	2001	−4.6	2002	−4.5	2003	−4.4	2004	−4.3	2005	−4.2	2006	−4.1	2007	−4.Ø	2008
−3.9	2009	−3.8	201Ø	−3.7	2011	−3.6	2012	−3.5	2013	−3.4	2014	−3.3	2015	−3.2	2016	−3.1	2017	−3.Ø	2018
−2.9	2019	−2.8	202Ø	−2.7	2021	−2.6	2022	−2.5	2023	−2.4	2024	−2.3	2025	−2.2	2026	−2.1	2027	−2.Ø	2028
−1.9	2029	−1.8	203Ø	−1.7	2031	−1.6	2032	−1.5	2033	−1.4	2034	−1.3	2035	−1.2	2036	−1.1	2037	−1.Ø	2038
−Ø.9	2039	−Ø.8	204Ø	−Ø.7	2041	−Ø.6	2042	−Ø.5	2043	−Ø.4	2044	−Ø.3	2045	−Ø.2	2046	−Ø.1	2047	Ø.Ø	2048
+Ø.1	2049	+Ø.2	205Ø	+Ø.3	2051	+Ø.4	2052	+Ø.5	2053	+Ø.6	2054	+Ø.7	2055	+Ø.8	2056	+Ø.9	2057	+1.Ø	2058
+1.1	2059	+1.2	206Ø	+1.3	2061	+1.4	2062	+1.5	2063	+1.6	2064	+1.7	2065	+1.8	2066	+1.9	2067	+2.Ø	2068
+2.1	2069	+2.2	207Ø	+2.3	2071	+2.4	2072	+2.5	2073	+2.6	2074	+2.7	2075	+2.8	2076	+2.9	2077	+3.Ø	2078
+3.1	2079	+3.2	208Ø	+3.3	2081	+3.4	2082	+3.5	2083	+3.6	2084	+3.7	2085	+3.8	2086	+3.9	2087	+4.Ø	2088
+4.1	2089	+4.2	209Ø	+4.3	2091	+4.4	2092	+4.5	2093	+4.6	2094	+4.7	2095	+4.8	2096	+4.9	2097	+5.Ø	2098
+5.1	2099	+5.2	210Ø	+5.3	2101	+5.4	2102	+5.5	2103	+5.6	2104	+5.7	2105	+5.8	2106	+5.9	2107	+6.Ø	2108
+6.1	2109	+6.2	211Ø	+6.3	2111	+6.4	2112	+6.5	2113	+6.6	2114	+6.7	2115	+6.8	2116	+6.9	2117	+7.Ø	2118
+7.1	2119	+7.2	212Ø	+7.3	2121	+7.4	2122	+7.5	2123	+7.6	2124	+7.7	2125	+7.8	2126	+7.9	2127	+8.Ø	2128
+8.1	2129	+8.2	213Ø	+8.3	2131	+8.4	2132	+8.5	2133	+8.6	2134	+8.7	2135	+8.8	2136	+8.9	2137	+9.Ø	2138
+9.1	2139	+9.2	214Ø	+9.3	2141	+9.4	2142	+9.5	2143	+9.6	2144	+9.7	2145	+9.8	2146	+9.9	2147	+1Ø.Ø	2148
+1Ø.1	2149	+1Ø.2	215Ø	+1Ø.3	2151	+1Ø.4	2152	+1Ø.5	2153	+1Ø.6	2154	+1Ø.7	2155	+1Ø.8	2156	+1Ø.9	2157	+11.Ø	2158
+11.1	2159	+11.2	216Ø	+11.3	2161	+11.4	2162	+11.5	2163	+11.6	2164	+11.7	2165	+11.8	2166	+11.9	2167	+12.Ø	2168
+12.1	2169	+12.2	217Ø	+12.3	2171	+12.4	2172	+12.5	2173	+12.6	2174	+12.7	2175	+12.8	2176	+12.9	2177	+13.Ø	2178
+13.1	2179	+13.2	218Ø	+13.3	2181	+13.4	2182	+13.5	2183	+13.6	2184	+13.7	2185	+13.8	2186	+13.9	2187	+14.Ø	2188
+14.1	2189	+14.2	219Ø	+14.3	2191	+14.4	2192	+14.5	2193	+14.6	2194	+14.7	2195	+14.8	2196	+14.9	2197	+15.Ø	2198
+15.1	2199	+15.2	220Ø	+15.3	2201	+15.4	2202	+15.5	2203	+15.6	2204	+15.7	2205	+15.8	2206	+15.9	2207	+16.Ø	2208
+16.1	2209	+16.2	221Ø	+16.3	2211	+16.4	2212	+16.5	2213	+16.6	2214	+16.7	2215	+16.8	2216	+16.9	2217	+17.Ø	2218
+17.1	2219	+17.2	222Ø	+17.3	2221	+17.4	2222	+17.5	2223	+17.6	2224	+17.7	2225	+17.8	2226	+17.9	2227	+18.Ø	2228
+18.1	2229	+18.2	223Ø	+18.3	2231	+18.4	2232	+18.5	2233	+18.6	2234	+18.7	2235	+18.8	2236	+18.9	2237	+19.Ø	2238
+19.1	2239	+19.2	224Ø	+19.3	2241	+19.4	2242	+19.5	2243	+19.6	2244	+19.7	2245	+19.8	2246	+19.9	2247	+2Ø.Ø	2248
+2Ø.1	2249	+2Ø.2	225Ø	+2Ø.3	2251	+2Ø.4	2252	+2Ø.5	2253	+2Ø.6	2254	+2Ø.7	2255	+2Ø.8	2256	+2Ø.9	2257	+21.Ø	2258
+21.1	2259	+21.2	226Ø	+21.3	2261	+21.4	2262	+21.5	2263	+21.6	2264	+21.7	2265	+21.8	2266	+21.9	2267	+22.Ø	2268
+22.1	2269	+22.2	227Ø	+22.3	2271	+22.4	2272	+22.5	2273	+22.6	2274	+22.7	2275	+22.8	2276	+22.9	2277	+23.Ø	2278
+23.1	2279	+23.2	228Ø	+23.3	2281	+23.4	2282	+23.5	2283	+23.6	2284	+23.7	2285	+23.8	2286	+23.9	2287	+24.Ø	2288

dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61
+24.1	2289	+24.2	229Ø	+24.3	2291	+24.4	2292	+24.5	2293	+24.6	2294	+24.7	2295	+24.8	2296	+24.9	2297	+25.Ø	2298
+25.1	2299	+25.2	230Ø	+25.3	2301	+25.4	2302	+25.5	2303	+25.6	2304	+25.7	2305	+25.8	2306	+25.9	2307	+26.Ø	2308
+26.1	2309	+26.2	231Ø	+26.3	2311	+26.4	2312	+26.5	2313	+26.6	2314	+26.7	2315	+26.8	2316	+26.9	2317	+27.Ø	2318
+27.1	2319	+27.2	232Ø	+27.3	2321	+27.4	2322	+27.5	2323	+27.6	2324	+27.7	2325	+27.8	2326	+27.9	2327	+28.Ø	2328
+28.1	2329	+28.2	233Ø	+28.3	2331	+28.4	2332	+28.5	2333	+28.6	2334	+28.7	2335	+28.8	2336	+28.9	2337	+29.Ø	2338
+29.1	2339	+29.2	234Ø	+29.3	2341	+29.4	2342	+29.5	2343	+29.6	2344	+29.7	2345	+29.8	2346	+29.9	2347	+3Ø.Ø	2348
+3Ø.1	2349	+3Ø.2	235Ø	+3Ø.3	2351	+3Ø.4	2352	+3Ø.5	2353	+3Ø.6	2354	+3Ø.7	2355	+3Ø.8	2356	+3Ø.9	2357	+31.Ø	2358
+31.1	2359	+31.2	236Ø	+31.3	2361	+31.4	2362	+31.5	2363	+31.6	2364	+31.7	2365	+31.8	2366	+31.9	2367	+32.Ø	2368
+32.1	2369	+32.2	237Ø	+32.3	2371	+32.4	2372	+32.5	2373	+32.6	2374	+32.7	2375	+32.8	2376	+32.9	2377	+33.Ø	2378
+33.1	2379	+33.2	238Ø	+33.3	2381	+33.4	2382	+33.5	2383	+33.6	2384	+33.7	2385	+33.8	2386	+33.9	2387	+34.Ø	2388
+34.1	2389	+34.2	239Ø	+34.3	2391	+34.4	2392	+34.5	2393	+34.6	2394	+34.7	2395	+34.8	2396	+34.9	2397	+35.Ø	2398
+35.1	2399	+35.2	24ØØ	+35.3	24Ø1	+35.4	24Ø2	+35.5	24Ø3	+35.6	24Ø4	+35.7	24Ø5	+35.8	24Ø6	+35.9	24Ø7	+36.Ø	24Ø8
+36.1	24Ø9	+36.2	241Ø	+36.3	2411	+36.4	2412	+36.5	2413	+36.6	2414	+36.7	2415	+36.8	2416	+36.9	2417	+37.Ø	2418
+37.1	2419	+37.2	242Ø	+37.3	2421	+37.4	2422	+37.5	2423	+37.6	2424	+37.7	2425	+37.8	2426	+37.9	2427	+38.Ø	2428
+38.1	2429	+38.2	243Ø	+38.3	2431	+38.4	2432	+38.5	2433	+38.6	2434	+38.7	2435	+38.8	2436	+38.9	2437	+39.Ø	2438
+39.1	2439	+39.2	244Ø	+39.3	2441	+39.4	2442	+39.5	2443	+39.6	2444	+39.7	2445	+39.8	2446	+39.9	2447	+4Ø.Ø	2448
+4Ø.1	2449	+4Ø.2	245Ø	+4Ø.3	2451	+4Ø.4	2452	+4Ø.5	2453	+4Ø.6	2454	+4Ø.7	2455	+4Ø.8	2456	+4Ø.9	2457	+41.Ø	2458
+41.1	2459	+41.2	246Ø	+41.3	2461	+41.4	2462	+41.5	2463	+41.6	2464	+41.7	2465	+41.8	2466	+41.9	2467	+42.Ø	2468
+42.1	2469	+42.2	247Ø	+42.3	2471	+42.4	2472	+42.5	2473	+42.6	2474	+42.7	2475	+42.8	2476	+42.9	2477	+43.Ø	2478
+43.1	2479	+43.2	248Ø	+43.3	2481	+43.4	2482	+43.5	2483	+43.6	2484	+43.7	2485	+43.8	2486	+43.9	2487	+44.Ø	2488
+44.1	2489	+44.2	249Ø	+44.3	2491	+44.4	2492	+44.5	2493	+44.6	2494	+44.7	2495	+44.8	2496	+44.9	2497	+45.Ø	2498
+45.1	2499	+45.2	25ØØ	+45.3	25Ø1	+45.4	25Ø2	+45.5	25Ø3	+45.6	25Ø4	+45.7	25Ø5	+45.8	25Ø6	+45.9	25Ø7	+46.Ø	25Ø8
+46.1	25Ø9	+46.2	251Ø	+46.3	2511	+46.4	2512	+46.5	2513	+46.6	2514	+46.7	2515	+46.8	2516	+46.9	2517	+47.Ø	2518
+47.1	2519	+47.2	252Ø	+47.3	2521	+47.4	2522	+47.5	2523	+47.6	2524	+47.7	2525	+47.8	2526	+47.9	2527	+48.Ø	2528
+48.1	2529	+48.2	253Ø	+48.3	2531	+48.4	2532	+48.5	2533	+48.6	2534	+48.7	2535	+48.8	2536	+48.9	2537	+49.Ø	2538
+49.1	2539	+49.2	254Ø	+49.3	2541	+49.4	2542	+49.5	2543	+49.6	2544	+49.7	2545	+49.8	2546	+49.9	2547	+5Ø.Ø	2548
+5Ø.1	2549	+5Ø.2	255Ø	+5Ø.3	2551	+5Ø.4	2552	+5Ø.5	2553	+5Ø.6	2554	+5Ø.7	2555	+5Ø.8	2556	+5Ø.9	2557	+51.Ø	2558
+51.1	2559	+51.2	256Ø	+51.3	2561	+51.4	2562	+51.5	2563	+51.6	2564	+51.7	2565	+51.8	2566	+51.9	2567	+52.Ø	2568
+52.1	2569	+52.2	257Ø	+52.3	2571	+52.4	2572	+52.5	2573	+52.6	2574	+52.7	2575	+52.8	2576	+52.9	2577	+53.Ø	2578
+53.1	2579	+53.2	258Ø	+53.3	2581	+53.4	2582	+53.5	2583	+53.6	2584	+53.7	2585	+53.8	2586	+53.9	2587	+54.Ø	2588
+54.1	2589	+54.2	259Ø	+54.3	2591	+54.4	2592	+54.5	2593	+54.6	2594	+54.7	2595	+54.8	2596	+54.9	2597	+55.Ø	2598
+55.1	2599	+55.2	26ØØ	+55.3	26Ø1	+55.4	26Ø2	+55.5	26Ø3	+55.6	26Ø4	+55.7	26Ø5	+55.8	26Ø6	+55.9	26Ø7	+56.Ø	26Ø8
+56.1	26Ø9	+56.2	261Ø	+56.3	2611	+56.4	2612	+56.5	2613	+56.6	2614	+56.7	2615	+56.8	2616	+56.9	2617	+57.Ø	2618

dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61	dB Value	X61
+57.1	2619	+57.2	262Ø	+57.3	2621	+57.4	2622	+57.5	2623	+57.6	2624	+57.7	2625	+57.8	2626	+57.9	2627	+58.Ø	2628
+58.1	2629	+58.2	263Ø	+58.3	2631	+58.4	2632	+58.5	2633	+58.6	2634	+58.7	2635	+58.8	2636	+58.9	2637	+59.Ø	2638
+59.1	2639	+59.2	264Ø	+59.3	2641	+59.4	2642	+59.5	2643	+59.6	2644	+59.7	2645	+59.8	2646	+59.9	2647	+6Ø.Ø	2648
+6Ø.1	2649	+6Ø.2	265Ø	+6Ø.3	2651	+6Ø.4	2652	+6Ø.5	2653	+6Ø.6	2654	+6Ø.7	2655	+6Ø.8	2656	+6Ø.9	2657	+61.Ø	2658
+61.1	2659	+61.2	266Ø	+61.3	2661	+61.4	2662	+61.5	2663	+61.6	2664	+61.7	2665	+61.8	2666	+61.9	2667	+62.Ø	2668
+62.1	2669	+62.2	267Ø	+62.3	2671	+62.4	2672	+62.5	2673	+62.6	2674	+62.7	2675	+62.8	2676	+62.9	2677	+63.Ø	2678
+63.1	2679	+63.2	268Ø	+63.3	2681	+63.4	2682	+63.5	2683	+63.6	2684	+63.7	2685	+63.8	2686	+63.9	2687	+64.Ø	2688
+64.1	2689	+64.2	269Ø	+64.3	2691	+64.4	2692	+64.5	2693	+64.6	2694	+64.7	2695	+64.8	2696	+64.9	2697	+65.Ø	2698
+65.1	2699	+65.2	27ØØ	+65.3	27Ø1	+65.4	27Ø2	+65.5	27Ø3	+65.6	27Ø4	+65.7	27Ø5	+65.8	27Ø6	+65.9	27Ø7	+66.Ø	27Ø8
+66.1	27Ø9	+66.2	271Ø	+66.3	2711	+66.4	2712	+66.5	2713	+66.6	2714	+66.7	2715	+66.8	2716	+66.9	2717	+67.Ø	2718
+67.1	2719	+67.2	272Ø	+67.3	2721	+67.4	2722	+67.5	2723	+67.6	2724	+67.7	2725	+67.8	2726	+67.9	2727	+68.Ø	2728
+68.1	2729	+68.2	273Ø	+68.3	2731	+68.4	2732	+68.5	2733	+68.6	2734	+68.7	2735	+68.8	2736	+68.9	2737	+69.Ø	2738
+69.1	2739	+69.2	274Ø	+69.3	2741	+69.4	2742	+69.5	2743	+69.6	2744	+69.7	2745	+69.8	2746	+69.9	2747	+7Ø.Ø	2748
+7Ø.1	2749	+7Ø.2	275Ø	+7Ø.3	2751	+7Ø.4	2752	+7Ø.5	2753	+7Ø.6	2754	+7Ø.7	2755	+7Ø.8	2756	+7Ø.9	2757	+71.Ø	2758
+71.1	2759	+71.2	276Ø	+71.3	2761	+71.4	2762	+71.5	2763	+71.6	2764	+71.7	2765	+71.8	2766	+71.9	2767	+72.Ø	2768
+72.1	2769	+72.2	277Ø	+72.3	2771	+72.4	2772	+72.5	2773	+72.6	2774	+72.7	2775	+72.8	2776	+72.9	2777	+73.Ø	2778
+73.1	2779	+73.2	278Ø	+73.3	2781	+73.4	2782	+73.5	2783	+73.6	2784	+73.7	2785	+73.8	2786	+73.9	2787	+74.Ø	2788
+74.1	2789	+74.2	279Ø	+74.3	2791	+74.4	2792	+74.5	2793	+74.6	2794	+74.7	2795	+74.8	2796	+74.9	2797	+75.Ø	2798
+75.1	2799	+75.2	28ØØ	+75.3	28Ø1	+75.4	28Ø2	+75.5	28Ø3	+75.6	28Ø4	+75.7	28Ø5	+75.8	28Ø6	+75.9	28Ø7	+76.Ø	28Ø8
+76.1	28Ø9	+76.2	281Ø	+76.3	2811	+76.4	2812	+76.5	2813	+76.6	2814	+76.7	2815	+76.8	2816	+76.9	2817	+77.Ø	2818
+77.1	2819	+77.2	282Ø	+77.3	2821	+77.4	2822	+77.5	2823	+77.6	2824	+77.7	2825	+77.8	2826	+77.9	2827	+78.Ø	2828
+78.1	2829	+78.2	283Ø	+78.3	2831	+78.4	2832	+78.5	2833	+78.6	2834	+78.7	2835	+78.8	2836	+78.9	2837	+79.Ø	2838
+79.1	2839	+79.2	284Ø	+79.3	2841	+79.4	2842	+79.5	2843	+79.6	2844	+79.7	2845	+79.8	2846	+79.9	2847	+8Ø.Ø	2848

HTML Operation

This section describes HTML operation and control of the DMP 128, including:

- **Download the Startup Page**
- **Status Tab**
- **Configuration Tab**
- **File Management Tab**
- **Special Characters**

The DMP 128 can be accessed by a PC or controller from a LAN or WAN using the Ethernet port and a web browser such as the Microsoft® Internet Explorer. The browser display of device status or operation has the appearance of web pages. This chapter describes the factory-installed HTML pages, which are always available and cannot be erased or overwritten.

NOTE: If the Ethernet connection to the device is unstable, try turning off the proxy server in the Web browser. In Microsoft Internet Explorer, click **Tools** > **Internet Options** > **Connections** > **LAN Settings**, uncheck the **"Use a proxy server..."** box, and then click **OK**.

Download the Startup Page

Access the device using HTML pages as follows:

1. Start the Web browser program.
2. Click in the browser **Address** field.
3. Enter the device IP address directly into the address field.

   NOTE: If the local system administrators have not changed the value, the factory-specified default IP address is 192.168.254.254.

4. If a custom display page is available, enter a slash (**/**) and the file name to open.

   NOTE: The browser address field should display the address in the following format: `xxx.xxx.xxx.xxx/{optional_file_name.HTML}`. The following characters are invalid in file names:
   {space} + ~ , @ = ‘ [ ] { } < > ’ " ; : > \ ?

5. Press <Enter>. The device checks to see if it is password protected.

   a. If the device is not password protected, it checks and downloads the HTML pages (proceed to step 7).
   b. If the device is password protected, the device opens the `Connect To` dialog box (see figure 85).

Figure 85. Connect To

6. Click in the **`Password`** field and type the appropriate administrator or user password. Click **`OK`**.

NOTE: A **`User Name`** entry is not required.

7. The device checks several possibilities, in the following order, then responds accordingly:

   a. Does the address include a specific file name, such as `10.13.156.10/file_name.HTML`? **If true**, the device downloads that HTML page.
   b. Is there a file in the device memory named "`index.HTML`"? **If true**, the device downloads "`index.HTML`" as the default startup page.
   c. **If neither of the above conditions is true**, the device downloads the factory-installed default startup page, "`nortxe_index.HTML`" (see figure 86 on the next page), also known as the System Status page.

Status Tab

System Status Page

The **System Status** page (see figure 86) provides an overall view of the status of the device, including system information, power supply status, and serial port settings. The **System Status** page is the default page when a connection to the device is established. Access the **System Status** page from other pages by clicking the **Status** tab.

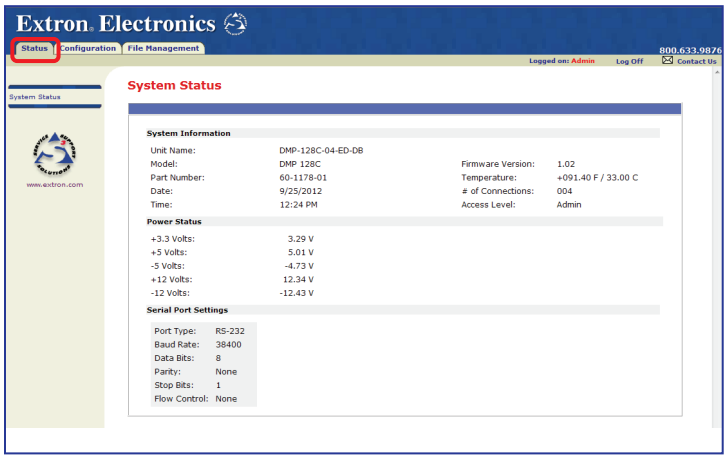


Figure 86. System Status Page

Configuration Tab

System Settings Page

Click the **Configuration** tab to open the **System Settings** page (see figure 87). The screen consists of fields to view and edit IP administration and system settings. **Passwords** and **Firmware Upgrade** pages are accessed by clicking the appropriate link on the left. See **Ethernet (LAN) Port** on page 130, for basic information about IP addresses and subnetting.

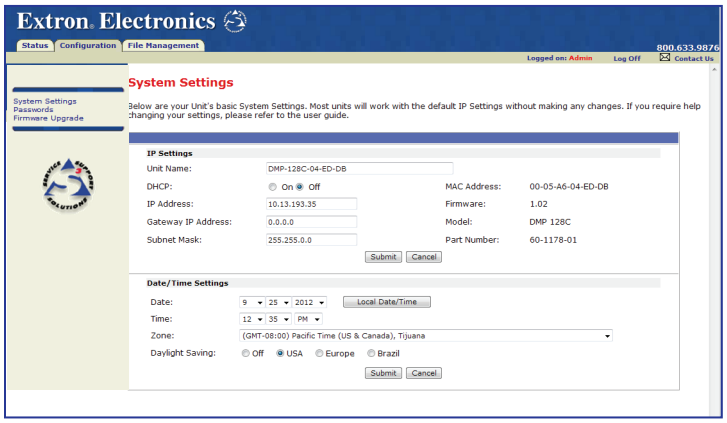


Figure 87. System Settings Page

On password-protected connections, there are two levels of protection: administrator and user. Administrators have full access to the passwords and firmware upgrade pages. Users have view only access.

- Ethernet connection to the device, either entering SIS commands (see **SIS Programming and Control** on page 129) or using the Extron DSP Configurator Program, (see **DSP Configurator Program Basics** on page 16), is password protected.
- Connection via any RS-232 port **is not** password protected.

IP Settings Panel

The **IP settings** fields provide a location for viewing and editing settings unique to the Ethernet interface. After editing any of the settings on this page, click **Submit** at the bottom of the section.

Unit Name Field

The **Unit Name** field contains the name of the device. This name field can be changed to any valid name, up to 24 alphanumeric characters.

NOTE: The following characters are invalid in the matrix name:
+ ~ , @ = ‘ [ ] { } < > ’ " ; : > \ and ?.

DHCP Radio Buttons

The **DHCP On** button directs the device to ignore any entered IP addresses and to obtain its IP address from a Dynamic Host Configuration Protocol (DHCP) server (if the network is DHCP capable). The **DHCP Off** button turns DHCP off. Contact the local system administrator for additional information on your network.

IP Address Field

The **IP Address** field contains the IP address encoded in the flash memory of the connected device.

Valid IP addresses consist of four 1-, 2-, or 3-digit numeric subfields separated by dots (periods). Each field can be numbered from **000** through 255. Leading zeroes, up to 3 digits total per field, are optional. Values of 256 and above are invalid.

The factory-installed default address is 192.168.254.254, but if this conflicts with other equipment at the installation site, change the IP address to any valid value.

NOTE: IP address changes can cause conflicts with other equipment. Only local system administrators should change IP addresses.

Gateway IP Address Field

The **Gateway IP Address** field identifies the address of the gateway to the mail server to be used if the device and the mail server are not on the same subnet.

The gateway IP address has the same validity rules as the system IP address.

Subnet Mask Field

The **Subnet Mask** field determines whether the device is on the same subnet as the mail server when you are subnetting.

MAC Address Field

The Media Access Control **(MAC) address** is hardcoded in the device and cannot be changed.

Firmware Field

The **Firmware** field displays the current firmware version being used by the device.

Model Field

The **Model** field displays the Extron model number of the device.

Part Number Field

The **Part Number** field displays the Extron Electronics part number of the device.

Date/Time Settings Panel

The **Date/Time Settings** fields (see figure 88) provide a location for viewing and setting the time functions.

Figure 88. Date/Time Settings Fields

Change the date and time settings as follows:

1. Click the desired drop-down list. Adjustable settings include month, day, year, hours, minutes, AM/PM, and (time) zone (the year is shown in figure 88).
2. If all variable selections are not visible, click and drag the slider or click the scroll up or scroll down button until the desired variable is visible.
3. Click the desired variable.

   NOTE: When setting the time, set the local time. The **Zone** variable allows you to select the offset from Greenwich Mean Time (GMT).

   The Zone field identifies the standard time zone selected and displays the amount of time, in hours and minutes, the local time varies from GMT international time reference.

4. Repeat steps 1 through 3 for other variables that require change.
5. If appropriate, click the **Daylight Savings** button to turn on the daylight savings time feature.

   NOTE: When Daylight Saving Time is on, the device automatically updates its internal clock between Standard Time and Daylight Saving Time in the spring and fall on the date the time change occurs in the country or region selected. When Daylight Saving Time is turned off, the device does not adjust its time reference.

6. Click **Submit**.

Passwords Page

Click the **Passwords** link to access the passwords page (see figure 89).

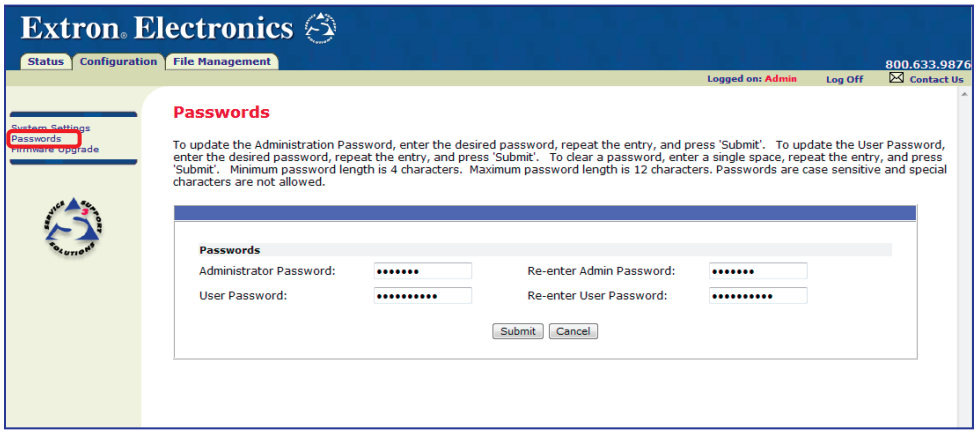

Extron Electronics
Status
Configuration
File Management
800.633.9876
Logged on: Admin
Log Off
Contact Us
System Settings
Passwords
Firmware Upgrade
Passwords
To update the Administration Password, enter the desired password, repeat the entry, and press 'Submit'. To update the User Password, enter the desired password, repeat the entry, and press 'Submit'. To clear a password, enter a single space, repeat the entry, and press 'Submit'. Minimum password length is 4 characters. Maximum password length is 12 characters. Passwords are case sensitive and special characters are not allowed.
Passwords
Administrator Password:
Re-enter Admin Password:
User Password:
Re-enter User Password:
Submit
Cancel

Figure 89. Passwords Page

The fields on the passwords page enter and verify administrator and user passwords. Passwords are case sensitive and limited to 12 upper case and lower case alphanumeric characters. Each password must be entered twice; once in the password field and then again in the **Re-enter Password** field. Characters in these fields are masked by asterisks (*****). If password protection is not desired, leave the password field and the Re-enter password field blank. After entering the desired password in both fields, click **Submit**.

NOTE: An administrator password must be created before a user password can be created.

Change a Password

To change a password, type the new password in both the **Password** and **Re-enter** password fields and click **Submit**.

Clear a Password

To clear an existing password so that no password is required, enter a single space in the password and re-enter password fields and click **Submit**.

Firmware Upgrade Page

NOTE: The DMP 128 firmware can be updated using the HTML pages or using a firmware loader utility. Extron recommends using the firmware loader utility available at **www.extron.com** (see **Firmware Loader** on page 168).

The Firmware Upgrade page provides another way to verify the current firmware version and to replace the firmware.

To update the device firmware using the HTML pages:

NOTE: The Firmware Upgrade page is **only** for replacing the firmware that controls device operation. To insert custom HTML pages, see **File Management Page** on page 164.

1. Visit the Extron Web site, **www.extron.com**, and select the **`Download Center`** tab.
2. Click the **`Firmware`** link (see figure 90).
3. Select the appropriate firmware file and click **`Download`**.
4. Enter the requested information.
5. Click **`Download`** to copy the firmware to your computer.

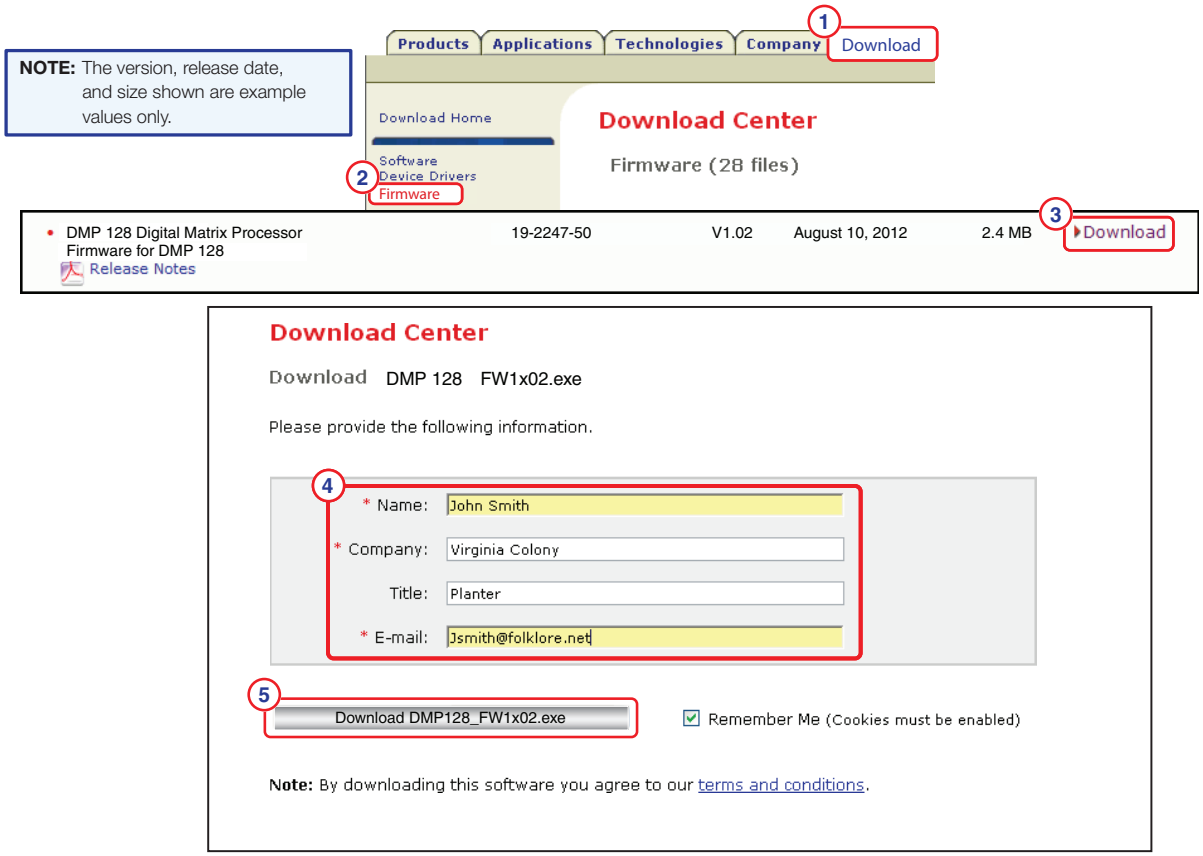


Figure 90. Location of Firmware Upgrade Files on the Web Site

6. Select **Run** twice (⑥ in figure 91). The PC downloads the firmware update from the Extron Web site and starts the installation program to extract the firmware file.

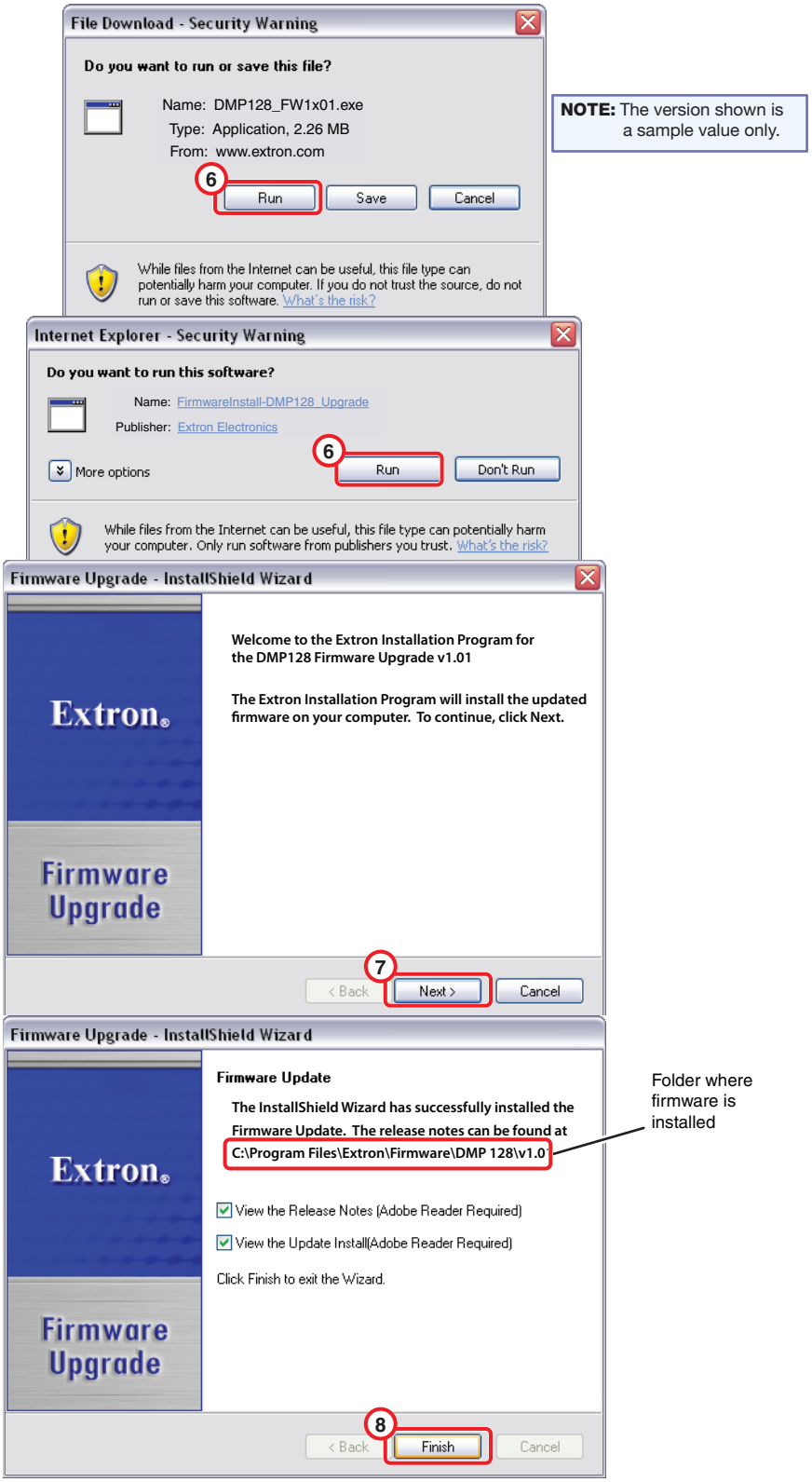

File Download - Security Warning
Do you want to run or save this file?
Name: DMP128_FW1x01.exe
Type: Application, 2.26 MB
From: www.extron.com
6
Run
Save
Cancel
While files from the Internet can be useful, this file type can potentially harm your computer. If you do not trust the source, do not run or save this software. What's the risk?
NOTE: The version shown is a sample value only.
Internet Explorer - Security Warning
Do you want to run this software?
Name: FirmwareInstall-DMP128_Upgrade
Publisher: Extron Electronics
More options
6
Run
Don't Run
While files from the Internet can be useful, this file type can potentially harm your computer. Only run software from publishers you trust. What's the risk?
Firmware Upgrade - InstallShield Wizard
Extron
Firmware Upgrade
Welcome to the Extron Installation Program for the DMP128 Firmware Upgrade v1.01
The Extron Installation Program will install the updated firmware on your computer. To continue, click Next.
7
< Back
Next >
Cancel
Firmware Upgrade - InstallShield Wizard
Extron
Firmware Upgrade
Firmware Update
The InstallShield Wizard has successfully installed the Firmware Update. The release notes can be found at
C:\Program Files\Extron\Firmware\DMP 128\v1.01
View the Release Notes (Adobe Reader Required)
View the Update Install(Adobe Reader Required)
Click Finish to exit the Wizard.
Folder where firmware is installed
8
< Back
Finish
Cancel

Figure 91. Downloading Firmware Upgrade Files

7. Click **Next** (see figure 91, ⑦ on the previous page). The program extracts and places the firmware files in a folder identified in the InstallShield Wizard dialog box.

   NOTE: The download folder can change depending on the operating system. Note the folder where the firmware file is saved.

8. Click **Finish** (figure 91, ⑧) to exit the program.
9. Connect the PC to the DMP 128 via the Ethernet port.
10. Access the device using the HTML pages (see **Download the Startup Page** on page 155).
11. Click the **Configuration** tab (see figure 92) to open the configuration page.
12. Click the **Firmware Upgrade** link on the System Configuration page.

    The current firmware version is displayed above the upload field for reference.

Figure 92. Firmware Upgrade Page

13. Click **Browse**. An Open File dialog box opens.
14. Navigate to the firmware file folder noted in step 7, and select the firmware file.

    NOTE: Valid firmware files must have the file extension '.S19'. Any other file extension is **not** a firmware upgrade. The original factory-installed firmware is permanently available on the device. If the attempted firmware upload fails for any reason, the device automatically reverts to the factory-installed firmware.

15. Click **Open**.
16. Click **Upload**. The firmware upload to the device can take a few minutes.

File Management Tab

File Management Page

To delete files such as HTML pages from the connected device or to upload custom files to the device, click the **File Management** tab. The device downloads the file management HTML page (see figure 93).

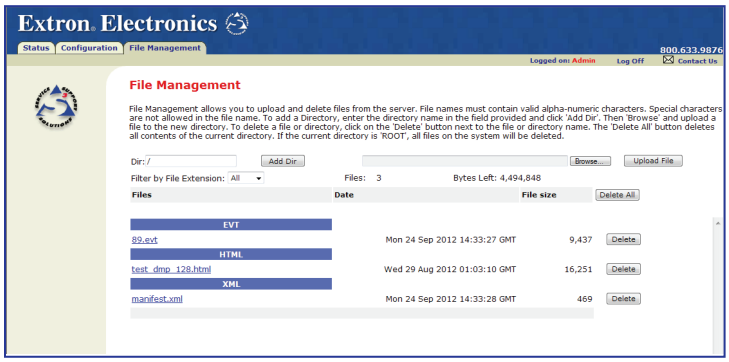

Extron Electronics
Status
Configuration
File Management
800.633.9876
Logged on: Admin
Log Off
Contact Us
File Management
File Management allows you to upload and delete files from the server. File names must contain valid alpha-numeric characters. Special characters are not allowed in the file name. To add a Directory, enter the directory name in the field provided and click 'Add Dir'. Then 'Browse' and upload a file to the new directory. To delete a file or directory, click on the 'Delete' button next to the file or directory name. The 'Delete All' button deletes all contents of the current directory. If the current directory is 'ROOT', all files on the system will be deleted.
Dir: /
Add Dir
Browse...
Upload File
Filter by File Extension: All
Files: 3
Bytes Left: 4,494,848
Files
Date
File size
Delete All
EVT
89.evt
Mon 24 Sep 2012 14:33:27 GMT
9,437
Delete
HTML
test_dmp_128.html
Wed 29 Aug 2012 01:03:10 GMT
16,251
Delete
XML
manifest.xml
Mon 24 Sep 2012 14:33:28 GMT
469
Delete

Figure 93. File Management Page

NOTE: The files listed in figure 93 are shown for example only.

To delete a file, click **Delete** at the right of that file.

Upload your own files as follows:

NOTE: The following characters are invalid in file names:
{space} + ~ , @ = ‘ [ ] { } < > ’ " ; : > \ ?

1. Click **Browse**.
2. Browse through the system and select the desired files.

   NOTE: If you want one of the pages that you create and upload to be the default startup page, name that file "**index.HTML**".

3. Click **Upload File**. The selected files appear in the list.

Special Characters

The HTML language reserves certain characters for specific functions. The device will not accept these characters as part of preset names, the name of the device, passwords, or locally created file names.

The device rejects the following characters:
{space} + ~ , @ = ‘ [ ] { } < > ’ “ ; : > \ ?

Reference Information

This section contains reference information for the DMP 128, including:

- **Mounting the DMP 128**
- **Firmware Loader**
- **DMP 128 Hardware Reset Modes**
- **DSP SIS Commands for Mix-point Level and Mute Control**
- **SIS Command Mix-points**

Mounting the DMP 128

The 1U high, full rack width, 8.5 inch deep DMP 128 Digital Matrix Processor can be:

- Set on a table,
- Mounted on a rack shelf,
- Mounted under a desk or tabletop.

Tabletop Use

Each DMP 128 is supplied with rubber feet (not installed). For tabletop use, attach a self-adhesive rubber foot to each corner of the bottom of the unit.

UL Rack Mounting Guidelines

The following Underwriters Laboratories (UL) guidelines pertain to the safe installation of the DMP 128 in a rack.

1. **Elevated operating ambient temperature** — If installed in a closed or multi-unit rack assembly, the operating ambient temperature of the rack environment may be greater than room ambient temperature. Therefore, install the unit in an environment compatible with the maximum ambient temperature (Tma = +122 °F, +50 °C) specified by Extron.
2. **Reduced air flow** — Install the equipment in a rack so that the amount of air flow required for safe operation of the equipment is not compromised.
3. **Mechanical loading** — Mount the equipment in the rack so that a hazardous condition is not achieved due to uneven mechanical loading.
4. **Circuit overloading** — Connect the equipment to the supply circuit and consider the effect that circuit overloading might have on overcurrent protection and supply wiring. Appropriate consideration of equipment nameplate ratings should be used when addressing this concern.
5. **Reliable earthing (grounding)** — Maintain reliable grounding of rack-mounted equipment. Pay particular attention to supply connections other than direct connections to the branch circuit (such as the use of power strips).

Rack Mounting

The DMP 128 is delivered with rack mounting brackets attached. For rack mounting, do not install the rubber feet. Use the rack ears to mount the DMP 128 in a standard equipment rack. The DMP 128 can be mounted on a 19 inch Universal 1U or Basic rack shelf.

To rack mount the DMP 128 on a rack shelf:

1. If rubber feet are installed on the bottom of the DMP 128, remove them.
2. Mount the DMP 128 on the rack shelf, using two 4-40 x 3/16 inch screws in opposite (diagonal) corners to secure the unit to the shelf. In a standard equipment rack, mount the DMP 128 using the installed rack mounting brackets.

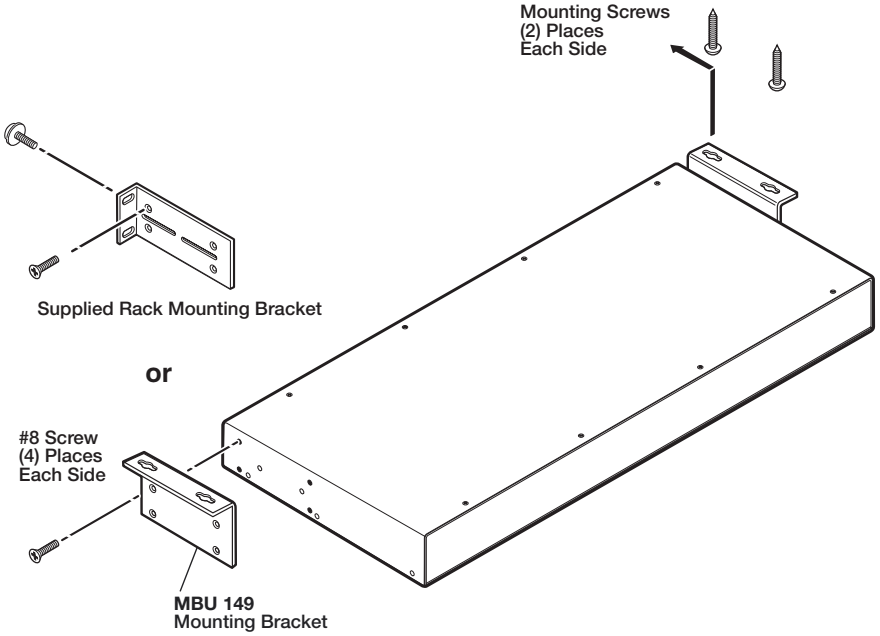


Figure 94. Mounting the DMP 128 on a Universal Rack Shelf

3. Install blank panels or other units on the rack shelf.

Table or Wall Mounting

The table or wall mounting brackets extend approximately 1/4 inch (6.4 mm) above the top surface of the enclosure. This design allows for an air space between the enclosure and the surface to which it is mounted.

Table or wall mount the DMP 128 as follows:

1. Remove the rack mounting brackets and attach the MBU 149 brackets in their place.
2. Hold the unit with the attached brackets against the underside of the table or other furniture, or against the wall. Mark the location of the screw holes of the bracket on the mounting surface.
3. Drill 3/32 inch (2 mm) diameter pilot holes, 1/4 inch (6.4 mm) deep in the mounting surface at the marked screw locations.
4. Insert #8 wood screws into the four pilot holes. Tighten each screw into the mounting surface until just less than 1/4 inch of the screw's head protrudes.
5. Align the mounting screws with the slots in the brackets and place the unit against the surface, with the screws through the bracket slots.
6. Slide the unit slightly forward or back, then tighten all four screws to secure it in place.

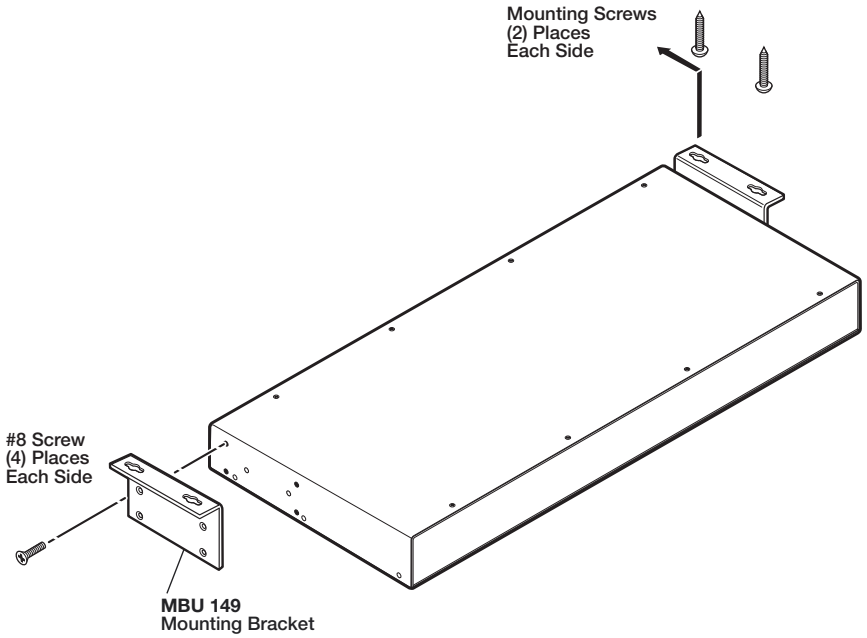


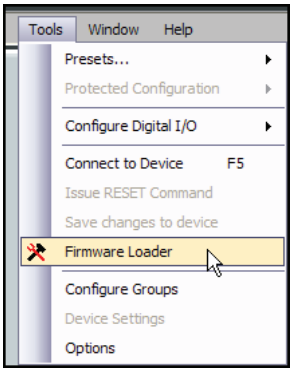
Figure 95. Table or Wall Mounting

Firmware Loader

Extron provides a firmware loader utility that, when installed, allows the firmware of the DSP Configurator program to be updated or replaced.

To install upgrade the DMP 128 firmware:

- Download, then install the Firmware Loader utility from **www.extron.com**.
- Download the desired firmware file from the Extron website, (see **Firmware Upgrade Page** on page 161).



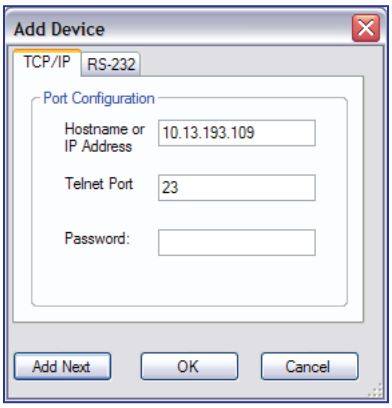
To access the firmware loader:

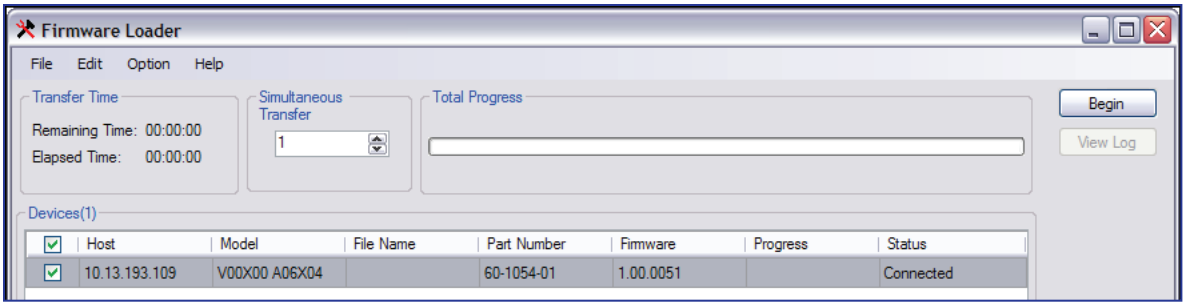
1. From the DSP Configurator toolbar, select **`Tools` > `Firmware Loader`**.

2. The `Add Device` dialog box opens. Type the IP address of the DMP 128 in the `Hostname or IP Address` field, then click **`OK`**.

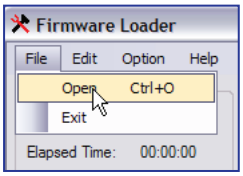
   NOTE: If the IP has not been changed, the default IP address is: 192.168.254.254

   The `Firmware Loader` screen opens.





3. From the toolbar, select **`File>Open`**.
4. Locate the downloaded firmware file and click it.
5. Click **`Begin`** on the main Firmware Loader dialog. The total progress bar tracks the loading progress.
6. When the upload is finished, exit the program by selecting **`File>Exit`**.



The firmware upload is complete.

DMP 128 Hardware Reset Modes

DMP 128 Reset Mode Summary				
	Mode	**Mode Activation**	**Result**	**Purpose/Notes**
Use Factory Firmware	**1**	Hold the reset button while applying power. **NOTE:** After a mode 1 reset, update the DMP 128 firmware to the latest version. DO NOT operate the firmware version that results from this mode reset.	The DMP 128 reverts to the factory default firmware. Event scripting does not start if the DMP 128 is powered on in this mode. All user files and settings (drivers, IP settings, and similar items) are maintained. **NOTE:** If you do not want to update the firmware or perform a mode 1 reset by mistake, cycle power to the DMP 128 to return to the firmware version running prior to the reset.	This mode reverts to the factory default firmware version if incompatibility issues arise with user-loaded firmware. **NOTE:** User-defined Web pages may not work correctly if using an earlier firmware version.
Run/Stop Events	**3**	With power on, press and hold the Reset button until the Power LED blinks once (3 sec.), then release and within 1 second press Reset momentarily (<1 sec). **NOTE:** The mode will not be entered unless the momentary press occurs within one second.	Mode 3 toggles events on or off. Front panel level indicators blink twice to indicate events has toggled on, or three times to indicate event logging has toggled off.	Useful for troubleshooting
Reset all IP Settings	**4**	Press and hold the Reset button for about 6 sec. until the Power LED blinks twice (once at 3 sec., again at 6 sec.), then release and within 1 second press Reset momentarily (< 1 sec.). **NOTE:** The mode will not be entered unless the momentary press occurs within one second.	**Mode 4:** • Enables ARP capability. • Set the IP address to default. • Sets the subnet to default. • Sets the gateway address to default. • Sets port mapping back to default. • Turns DHCP off. • Turns events off.	Enables resetting IP address information using ARP and MAC address.
Reset to Factory Defaults	**5**	Press and hold the Reset button for about 9 sec. until the Power LED blinks three times (once at 3 sec., again at 6 sec., again at 9 sec.), then release and within 1 second press Reset momentarily (< 1 sec.). **NOTE:** The mode will not be entered unless the momentary press occurs within one second.	**Mode 5 performs a complete reset to factory defaults, except for firmware:** • Does everything mode 4 reset does. • All mix-points are muted and set to 0 dB. • All outputs are unmuted and set to 0 dB. • DSP Processing returned to defaults and bypassed. • All inputs are muted and set to 0 dB. • All presets and group master memory cleared.	Useful to start over with configuration or uploading, and to replace events.

DSP SIS Commands for Mix-point Level and Mute Control

Many functions; gain, mute, group masters, mix-points, and a protected configuration can be controlled using SIS commands. These commands follow the same general rules as basic SIS commands, but the variables (Xn) tend to be more complex. Before attempting SIS control of the DSP, a comprehensive understanding of the audio signal flow is helpful to understanding the commands. Figure 96 shows the specific DSP functions available for SIS command control of the mix-points.

NOTE: This supplement refers only to the mix-point and corresponding level and mute control commands. See **DSP SIS Commands** on page 141 for basic SIS commands and functions.

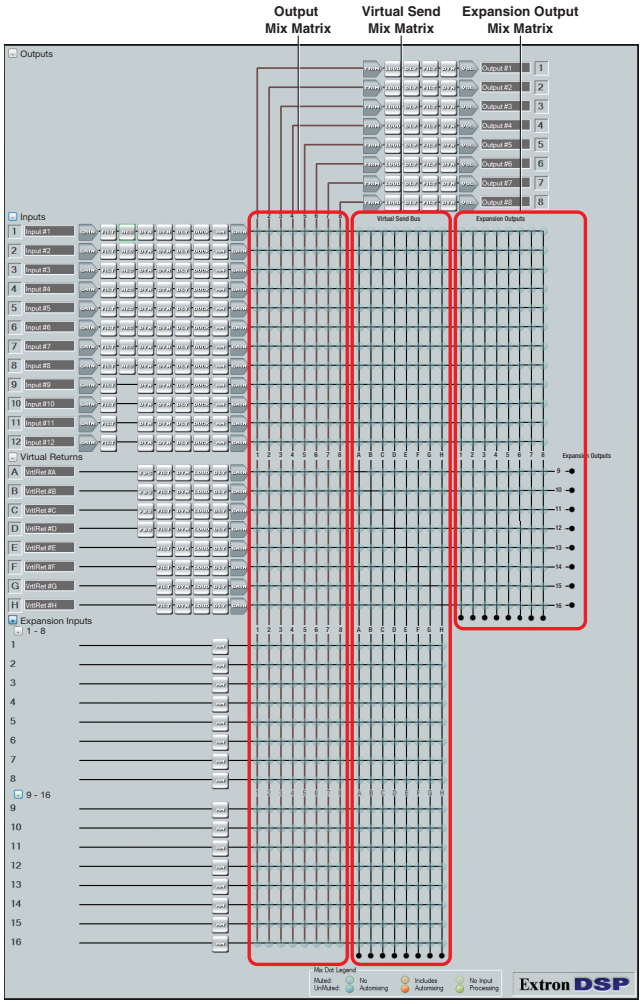


Figure 96. DSP Processors Addressable via SIS Commands

The Mix Matrix mix-points addressed by this section are:

① Output Mix Matrix (including level and mute)

② Virtual Send Mix Matrix (including level and mute)

③ Expansion Output Mix Matrix (including level and mute)

Setting the level of a single mix-point in the mix matrix requires an SIS command that contains both the mix-point location and level. Making or breaking a connection between a mix-point input and output requires the mix-point location be unmuted or muted.

Selecting Mix-points

Mix-points are controlled by their address. Each mix-point in figure 94 has an individual address selected and controlled using SIS commands. (A mix-point contains a level control and a routing control that mutes or unmutes the signal path from the input mix-point to the output.)

For example, to view the current status of the input 1 to output 1 mix-point (SIS address: 20000) enter:

Esc M X60 *AU← or Esc M20000*AU←

The response from the DMP 128 is:

DsM20000*1AU← if the mix-point is muted, or

Esc M20000*0AU↵ if the mix-point is unmuted, connecting the mix-point input with its output.

To select a mix-point to view its current signal level setting:

Esc G X60 AU←.

If looking at the same input 1 to output 1 mix-point, enter:

Esc G20000AU← or

where 20000 is the mix-point. The response is:

DsG20000*1948↵ .

Using the **table 2** cross-reference on page 149, the mix-point is set to -10 dB.

Signal Routing

Mix-point routing is achieved by unmuting the mix-point. To select a mix-point, use the tables starting on page 175 to find a specific mix-point. In this example, input 1 is routed to output 1.

Connecting a mix-point input to an output requires the mix-point be unmuted. Using the formula from above where X60 is the mix-point, to unmute any processor or mix-point use the command:

Esc M X60 *0AU←.

The input 1 to output 1 mix-point address according to table 3 on page 175 is 20000 making the correct formula Esc M20000*0AU←. This routes input 1 to output 1.

To break the signal, use the mute formula:

Esc M X60 *1AU← or Esc M20000*1AU←.

In this manner any mix-point can be selected and a connection made between its input and output, or if a connection has already been made, it can be broken.

Setting Audio Levels

The mix-point level is controlled using SIS commands. The commands template follows a similar structure as mute and unmute commands.

The audio conversion tables are the same for all mix-points (see **table 2** beginning on page 149). Although a greater range is shown in the table, the minimum and maximum levels are the same for all mix-points (–35.0 to +25.0 dB).

To select the mix-point, use the table 3 beginning on page 175 to find the specific mix-point. In this example, the audio level of mix-point input 1 to output 1 is set to -10 dB.

The audio level control and mix-point selection formula is:

Esc G X60 * X61 AU ←

where X60 is the mix-point location and X61 is the signal level. The mix-point location according to table 3 on page 175 is **2ØØØØ** and the SIS conversion for -10 dB from table 2 on page 149 is 1948 making the SIS command Esc G**2ØØØØ***1948AU. This sets the mix-point level at input 1 to output 1 to -10 dB.

To return the signal level back to default, use the same formula:

Esc G**2ØØØØ***2048AU sets the level back to **Ø** dB.

In this manner any mix-point can be selected and the signal level between its input and output changed within the range of - 35 to +25 dB.

SIS Command and Response Table for DSP Mix-point and Level Control

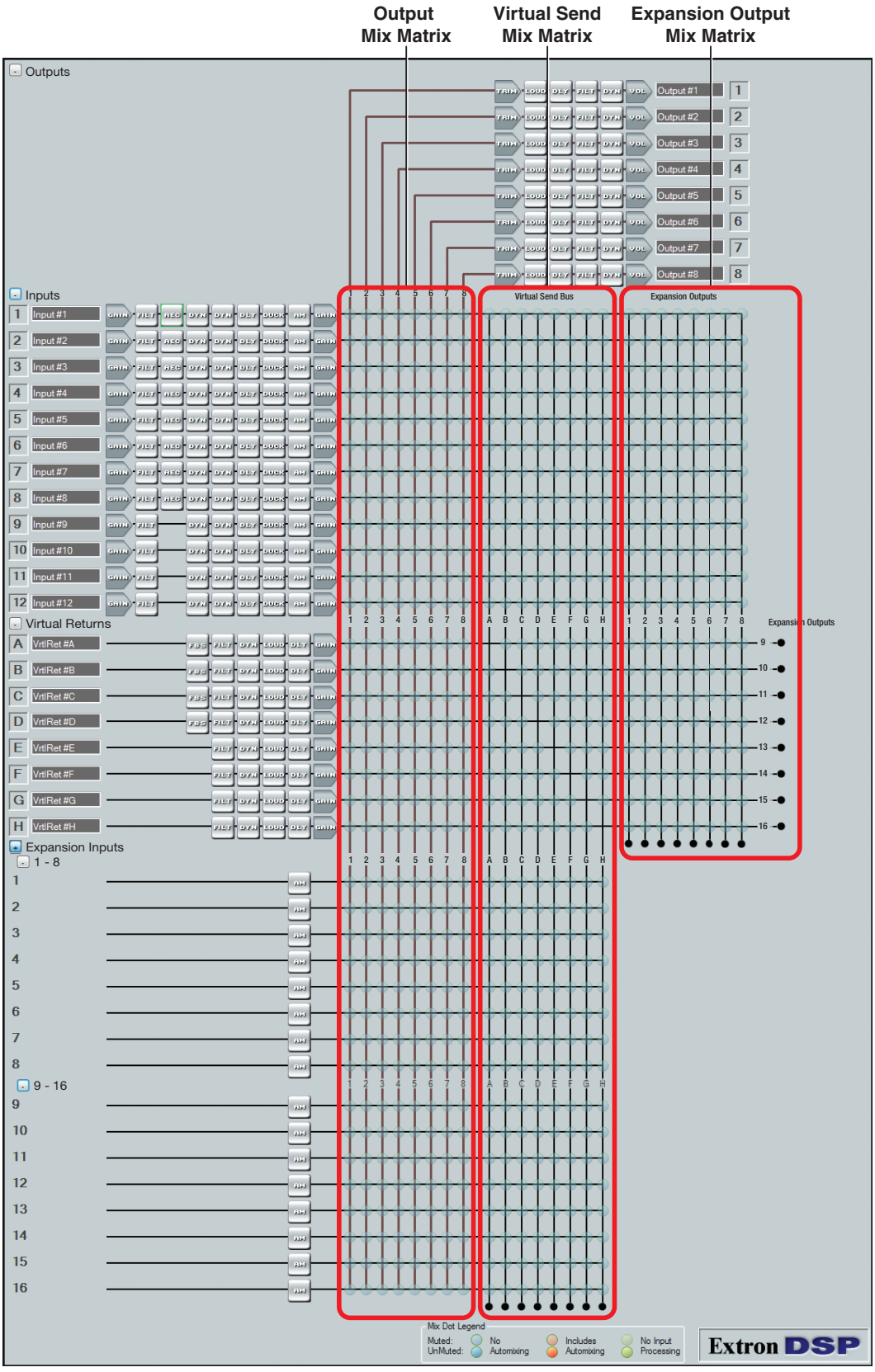
Setting the level of a mix-point requires the mix-point address and level value. Making a connection between a mix-point input and output requires the mix-point be unmuted.

Command	ASCII command (host to device)	Response (device to host)	Additional description
Audio level control, and mix-point selection			
NOTE: The command format is the same, regardless of the control or mix-point to be set; the acceptable adjustment range of the main mix-points is -35 dB to +25 dB, in Ø.1 dB increments. All responses are shown with the DMP 128 in Verbose mode 2 or 3.			
Set a mixpoint trim or gain	Esc G X60 * X61 AU ←	Ds G X60 * X61 ↵	Set trim or mix control X60 to a value of X61 dB.
Example 1 (Input 1 to Virtual Send A gain):	Esc G2ØØØ9*2Ø4ØAU ←	DsG2ØØØ9*2Ø4Ø↵	Set the In 1 to Virtual Send A gain to a value of -0.8 dB.
Example 2 (mix-point gain):	Esc G2ØØØ1*2213AU ←	DsG2ØØØ1*2213↵	Mix +16.5 dB of input 1 into output 2.
Read a mix-point level	Esc G X60 AU ←	Ds G X60 * X61 ↵	Mixpoint X60 is set to a value of X61 dB.
Example 1 Input 11 to Output 8:	Esc G21Ø7AU ←	DsG21Ø7*21Ø3↵	Mix-point In 11 to Output 8 is set to a value of +5.5 dB.
Example 2 Input 3 to Output 4:	Esc G2Ø2Ø3AU ←	DsG2Ø2Ø3*214Ø↵	+9.2 dB of In 3 is mixed to output 4.
Audio mute			
NOTE: All responses are shown with the mixer device in Verbose mode 2 or 3.			
Audio mute	Esc M X60 *1AU ←	DsM X60 *1↵	Mute mix-point X60.
Example: Input 4 to Output 2:	Esc M2Ø3Ø1*1AU ←	DsM2Ø3Ø1*1↵	Mute mix-point input 4 to output 2.
Audio unmute	Esc M X60 *ØAU ←	DsM X60 *Ø↵	Unmute mix-point X60.
Read mute status	Esc M X60 AU ←	DsM X60 * X64 ↵	X64: Ø = mute off (unmute), 1 = mute on (mute).

NOTES:			
X60	=	Mix-point select	See tables 3-10 on pages 175-197.
X61	=	Signal level value in dB	See table 2 on page 148.
X64	=	Mute status	Ø = unmute 1 = mute

SIS Command Mix-points

The mix-points are divided into three main matrix mixers: the Main or Output mix matrix (①), the virtual send bus mix matrix (②), and the expansion (EXP) outputs mix matrix (③).



Output Mixer

The output mixer routes the mic/line inputs (①), virtual returns (②), and expansion bus inputs (③), to the outputs.

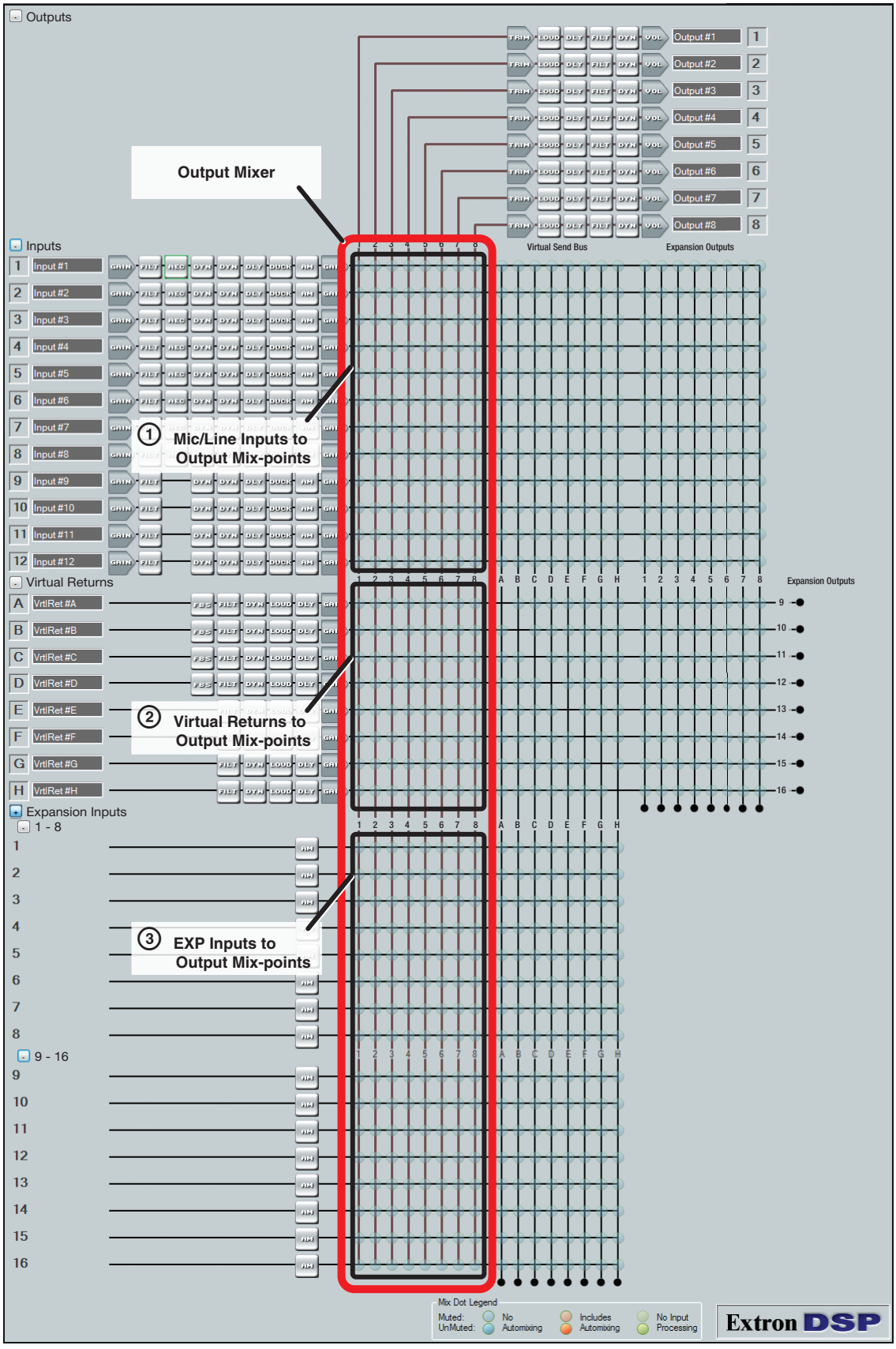


Table 3: Mic/Line Inputs to Output Mix-points (①)

① Mic/Line Inputs to Outputs	X60	① Mic/Line Inputs to Outputs	X60
Mic/Line Input 1 to Output 1	2ØØØØ	Mic/Line Input 2 to Output 1	2Ø1ØØ
Mic/Line Input 1 to Output 2	2ØØØ1	Mic/Line Input 2 to Output 2	2Ø1Ø1
Mic/Line Input 1 to Output 3	2ØØØ2	Mic/Line Input 2 to Output 3	2Ø1Ø2
Mic/Line Input 1 to Output 4	2ØØØ3	Mic/Line Input 2 to Output 4	2Ø1Ø3
Mic/Line Input 1 to Output 5	2ØØØ4	Mic/Line Input 2 to Output 5	2Ø1Ø4
Mic/Line Input 1 to Output 6	2ØØØ5	Mic/Line Input 2 to Output 6	2Ø1Ø5
Mic/Line Input 1 to Output 7	2ØØØ6	Mic/Line Input 2 to Output 7	2Ø1Ø6
Mic/Line Input 1 to Output 8	2ØØØ7	Mic/Line Input 2 to Output 8	2Ø1Ø7
Mic/Line Input 3 to Output 1	2Ø2ØØ	Mic/Line Input 4 to Output 1	2Ø3ØØ
Mic/Line Input 3 to Output 2	2Ø2Ø1	Mic/Line Input 4 to Output 2	2Ø3Ø1
Mic/Line Input 3 to Output 3	2Ø2Ø2	Mic/Line Input 4 to Output 3	2Ø3Ø2
Mic/Line Input 3 to Output 4	2Ø2Ø3	Mic/Line Input 4 to Output 4	2Ø3Ø3
Mic/Line Input 3 to Output 5	2Ø2Ø4	Mic/Line Input 4 to Output 5	2Ø3Ø4
Mic/Line Input 3 to Output 6	2Ø2Ø5	Mic/Line Input 4 to Output 6	2Ø3Ø5
Mic/Line Input 3 to Output 7	2Ø2Ø6	Mic/Line Input 4 to Output 7	2Ø3Ø6
Mic/Line Input 3 to Output 8	2Ø2Ø7	Mic/Line Input 4 to Output 8	2Ø3Ø7
Mic/Line Input 5 to Output 1	2Ø4ØØ	Mic/Line Input 6 to Output 1	2Ø5ØØ
Mic/Line Input 5 to Output 2	2Ø4Ø1	Mic/Line Input 6 to Output 2	2Ø5Ø1
Mic/Line Input 5 to Output 3	2Ø4Ø2	Mic/Line Input 6 to Output 3	2Ø5Ø2
Mic/Line Input 5 to Output 4	2Ø4Ø3	Mic/Line Input 6 to Output 4	2Ø5Ø3
Mic/Line Input 5 to Output 5	2Ø4Ø4	Mic/Line Input 6 to Output 5	2Ø5Ø4
Mic/Line Input 5 to Output 6	2Ø4Ø5	Mic/Line Input 6 to Output 6	2Ø5Ø5
Mic/Line Input 5 to Output 7	2Ø4Ø6	Mic/Line Input 6 to Output 7	2Ø5Ø6
Mic/Line Input 5 to Output 8	2Ø4Ø7	Mic/Line Input 6 to Output 8	2Ø5Ø7
Mic/Line Input 7 to Output 1	2Ø6ØØ	Mic/Line Input 8 to Output 1	2Ø7ØØ
Mic/Line Input 7 to Output 2	2Ø6Ø1	Mic/Line Input 8 to Output 2	2Ø7Ø1
Mic/Line Input 7 to Output 3	2Ø6Ø2	Mic/Line Input 8 to Output 3	2Ø7Ø2
Mic/Line Input 7 to Output 4	2Ø6Ø3	Mic/Line Input 8 to Output 4	2Ø7Ø3
Mic/Line Input 7 to Output 5	2Ø6Ø4	Mic/Line Input 8 to Output 5	2Ø7Ø4
Mic/Line Input 7 to Output 6	2Ø6Ø5	Mic/Line Input 8 to Output 6	2Ø7Ø5
Mic/Line Input 7 to Output 7	2Ø6Ø6	Mic/Line Input 8 to Output 7	2Ø7Ø6
Mic/Line Input 7 to Output 8	2Ø6Ø7	Mic/Line Input 8 to Output 8	2Ø7Ø7
Mic/Line Input 9 to Output 1	2Ø8ØØ	Mic/Line Input 10 to Output 1	2Ø9ØØ
Mic/Line Input 9 to Output 2	2Ø8Ø1	Mic/Line Input 10 to Output 2	2Ø9Ø1
Mic/Line Input 9 to Output 3	2Ø8Ø2	Mic/Line Input 10 to Output 3	2Ø9Ø2
Mic/Line Input 9 to Output 4	2Ø8Ø3	Mic/Line Input 10 to Output 4	2Ø9Ø3
Mic/Line Input 9 to Output 5	2Ø8Ø4	Mic/Line Input 10 to Output 5	2Ø9Ø4
Mic/Line Input 9 to Output 6	2Ø8Ø5	Mic/Line Input 10 to Output 6	2Ø9Ø5
Mic/Line Input 9 to Output 7	2Ø8Ø6	Mic/Line Input 10 to Output 7	2Ø9Ø6
Mic/Line Input 9 to Output 8	2Ø8Ø7	Mic/Line Input 10 to Output 8	2Ø9Ø7

① Mic/Line Inputs to Outputs	X60	① Mic/Line Inputs to Outputs	X60
Mic/Line Input 11 to Output 1	21ØØØ	Mic/Line Input 12 to Output 1	211ØØ
Mic/Line Input 11 to Output 2	21ØØ1	Mic/Line Input 12 to Output 2	211Ø1
Mic/Line Input 11 to Output 3	21ØØ2	Mic/Line Input 12 to Output 3	211Ø2
Mic/Line Input 11 to Output 4	21ØØ3	Mic/Line Input 12 to Output 4	211Ø3
Mic/Line Input 11 to Output 5	21ØØ4	Mic/Line Input 12 to Output 5	211Ø4
Mic/Line Input 11 to Output 6	21ØØ5	Mic/Line Input 12 to Output 6	211Ø5
Mic/Line Input 11 to Output 7	21ØØ6	Mic/Line Input 12 to Output 7	211Ø6
Mic/Line Input 11 to Output 8	21ØØ7	Mic/Line Input 12 to Output 8	211Ø7

Telephone Rx to Output 1	212ØØ
Telephone Rx to Output 2	212Ø1
Telephone Rx to Output 3	212Ø2
Telephone Rx to Output 4	212Ø3
Telephone Rx to Output 5	212Ø4
Telephone Rx to Output 6	212Ø5
Telephone Rx to Output 7	212Ø6
Telephone Rx to Output 8	212Ø7

Table 4: Virtual Returns to Output Mix-points (②)

② Virtual Returns to Outputs	X60	② Virtual Returns to Outputs	X60
Virtual Return A to Output 1	213ØØ	Virtual Return B to Output 1	214ØØ
Virtual Return A to Output 2	213Ø1	Virtual Return B to Output 2	214Ø1
Virtual Return A to Output 3	213Ø2	Virtual Return B to Output 3	214Ø2
Virtual Return A to Output 4	213Ø3	Virtual Return B to Output 4	214Ø3
Virtual Return A to Output 5	213Ø4	Virtual Return B to Output 5	214Ø4
Virtual Return A to Output 6	213Ø5	Virtual Return B to Output 6	214Ø5
Virtual Return A to Output 7	213Ø6	Virtual Return B to Output 7	214Ø6
Virtual Return A to Output 8	213Ø7	Virtual Return B to Output 8	214Ø7
Virtual Return C to Output 1	215ØØ	Virtual Return D to Output 1	216ØØ
Virtual Return C to Output 2	215Ø1	Virtual Return D to Output 2	216Ø1
Virtual Return C to Output 3	215Ø2	Virtual Return D to Output 3	216Ø2
Virtual Return C to Output 4	215Ø3	Virtual Return D to Output 4	216Ø3
Virtual Return C to Output 5	215Ø4	Virtual Return D to Output 5	216Ø4
Virtual Return C to Output 6	215Ø5	Virtual Return D to Output 6	216Ø5
Virtual Return C to Output 7	215Ø6	Virtual Return D to Output 7	216Ø6
Virtual Return C to Output 8	215Ø7	Virtual Return D to Output 8	216Ø7
Virtual Return E to Output 1	217ØØ	Virtual Return F to Output 1	218ØØ
Virtual Return E to Output 2	217Ø1	Virtual Return F to Output 2	218Ø1
Virtual Return E to Output 3	217Ø2	Virtual Return F to Output 3	218Ø2
Virtual Return E to Output 4	217Ø3	Virtual Return F to Output 4	218Ø3
Virtual Return E to Output 5	217Ø4	Virtual Return F to Output 5	218Ø4
Virtual Return E to Output 6	217Ø5	Virtual Return F to Output 6	218Ø5
Virtual Return E to Output 7	217Ø6	Virtual Return F to Output 7	218Ø6
Virtual Return E to Output 8	217Ø7	Virtual Return F to Output 8	218Ø7
Virtual Return G to Output 1	219ØØ	Virtual Return H to Output 1	22ØØØ
Virtual Return G to Output 2	219Ø1	Virtual Return H to Output 2	22ØØ1
Virtual Return G to Output 3	219Ø2	Virtual Return H to Output 3	22ØØ2
Virtual Return G to Output 4	219Ø3	Virtual Return H to Output 4	22ØØ3
Virtual Return G to Output 5	219Ø4	Virtual Return H to Output 5	22ØØ4
Virtual Return G to Output 6	219Ø5	Virtual Return H to Output 6	22ØØ5
Virtual Return G to Output 7	219Ø6	Virtual Return H to Output 7	22ØØ6
Virtual Return G to Output 8	219Ø7	Virtual Return H to Output 8	22ØØ7

Table 5: EXP Inputs to Output Mix-points (③)

③ EXP Returns to Outputs	X60	③ EXP Returns to Outputs	X60
EXP Input 1 to Output 1	221ØØ	EXP Input 2 to Output 1	222ØØ
EXP Input 1 to Output 2	221Ø1	EXP Input 2 to Output 2	222Ø1
EXP Input 1 to Output 3	221Ø2	EXP Input 2 to Output 3	222Ø2
EXP Input 1 to Output 4	221Ø3	EXP Input 2 to Output 4	222Ø3
EXP Input 1 to Output 5	221Ø4	EXP Input 2 to Output 5	222Ø4
EXP Input 1 to Output 6	221Ø5	EXP Input 2 to Output 6	222Ø5
EXP Input 1 to Output 7	221Ø6	EXP Input 2 to Output 7	222Ø6
EXP Input 1 to Output 8	221Ø7	EXP Input 2 to Output 8	222Ø7
EXP Input 3 to Output 1	223ØØ	EXP Input 4 to Output 1	224ØØ
EXP Input 3 to Output 2	223Ø1	EXP Input 4 to Output 2	224Ø1
EXP Input 3 to Output 3	223Ø2	EXP Input 4 to Output 3	224Ø2
EXP Input 3 to Output 4	223Ø3	EXP Input 4 to Output 4	224Ø3
EXP Input 3 to Output 5	223Ø4	EXP Input 4 to Output 5	224Ø4
EXP Input 3 to Output 6	223Ø5	EXP Input 4 to Output 6	224Ø5
EXP Input 3 to Output 7	223Ø6	EXP Input 4 to Output 7	224Ø6
EXP Input 3 to Output 8	223Ø7	EXP Input 4 to Output 8	224Ø7
EXP Input 5 to Output 1	225ØØ	EXP Input 6 to Output 1	226ØØ
EXP Input 5 to Output 2	225Ø1	EXP Input 6 to Output 2	226Ø1
EXP Input 5 to Output 3	225Ø2	EXP Input 6 to Output 3	226Ø2
EXP Input 5 to Output 4	225Ø3	EXP Input 6 to Output 4	226Ø3
EXP Input 5 to Output 5	225Ø4	EXP Input 6 to Output 5	226Ø4
EXP Input 5 to Output 6	225Ø5	EXP Input 6 to Output 6	226Ø5
EXP Input 5 to Output 7	225Ø6	EXP Input 6 to Output 7	226Ø6
EXP Input 5 to Output 8	225Ø7	EXP Input 6 to Output 8	226Ø7
EXP Input 7 to Output 1	227ØØ	EXP Input 8 to Output 1	228ØØ
EXP Input 7 to Output 2	227Ø1	EXP Input 8 to Output 2	228Ø1
EXP Input 7 to Output 3	227Ø2	EXP Input 8 to Output 3	228Ø2
EXP Input 7 to Output 4	227Ø3	EXP Input 8 to Output 4	228Ø3
EXP Input 7 to Output 5	227Ø4	EXP Input 8 to Output 5	228Ø4
EXP Input 7 to Output 6	227Ø5	EXP Input 8 to Output 6	228Ø5
EXP Input 7 to Output 7	227Ø6	EXP Input 8 to Output 7	228Ø6
EXP Input 7 to Output 8	227Ø7	EXP Input 8 to Output 8	228Ø7
EXP Input 9 to Output 1	229ØØ	EXP Input 10 to Output 1	23ØØØ
EXP Input 9 to Output 2	229Ø1	EXP Input 10 to Output 2	23ØØ1
EXP Input 9 to Output 3	229Ø2	EXP Input 10 to Output 3	23ØØ2
EXP Input 9 to Output 4	229Ø3	EXP Input 10 to Output 4	23ØØ3
EXP Input 9 to Output 5	229Ø4	EXP Input 10 to Output 5	23ØØ4
EXP Input 9 to Output 6	229Ø5	EXP Input 10 to Output 6	23ØØ5
EXP Input 9 to Output 7	229Ø6	EXP Input 10 to Output 7	23ØØ6
EXP Input 9 to Output 8	229Ø7	EXP Input 10 to Output 8	23ØØ7

③ EXP Returns to Outputs	X60	③ EXP Returns to Outputs	X60
EXP Input 11 to Output 1	231ØØ	EXP Input 12 to Output 1	232ØØ
EXP Input 11 to Output 2	231Ø1	EXP Input 12 to Output 2	232Ø1
EXP Input 11 to Output 3	231Ø2	EXP Input 12 to Output 3	232Ø2
EXP Input 11 to Output 4	231Ø3	EXP Input 12 to Output 4	232Ø3
EXP Input 11 to Output 5	231Ø4	EXP Input 12 to Output 5	232Ø4
EXP Input 11 to Output 6	231Ø5	EXP Input 12 to Output 6	232Ø5
EXP Input 11 to Output 7	231Ø6	EXP Input 12 to Output 7	232Ø6
EXP Input 11 to Output 8	231Ø7	EXP Input 12 to Output 8	232Ø7
EXP Input 13 to Output 1	233ØØ	EXP Input 14 to Output 1	234ØØ
EXP Input 13 to Output 2	233Ø1	EXP Input 14 to Output 2	234Ø1
EXP Input 13 to Output 3	233Ø2	EXP Input 14 to Output 3	234Ø2
EXP Input 13 to Output 4	233Ø3	EXP Input 14 to Output 4	234Ø3
EXP Input 13 to Output 5	233Ø4	EXP Input 14 to Output 5	234Ø4
EXP Input 13 to Output 6	233Ø5	EXP Input 14 to Output 6	234Ø5
EXP Input 13 to Output 7	233Ø6	EXP Input 14 to Output 7	234Ø6
EXP Input 13 to Output 8	233Ø7	EXP Input 14 to Output 8	234Ø7
EXP Input 15 to Output 1	235ØØ	EXP Input 16 to Output 1	236ØØ
EXP Input 15 to Output 2	235Ø1	EXP Input 16 to Output 2	236Ø1
EXP Input 15 to Output 3	235Ø2	EXP Input 16 to Output 3	236Ø2
EXP Input 15 to Output 4	235Ø3	EXP Input 16 to Output 4	236Ø3
EXP Input 15 to Output 5	235Ø4	EXP Input 16 to Output 5	236Ø4
EXP Input 15 to Output 6	235Ø5	EXP Input 16 to Output 6	236Ø5
EXP Input 15 to Output 7	235Ø6	EXP Input 16 to Output 7	236Ø6
EXP Input 15 to Output 8	235Ø7	EXP Input 16 to Output 8	236Ø7
EXP Input 17 to Output 1	237ØØ	EXP Input 18 to Output 1	238ØØ
EXP Input 17 to Output 2	237Ø1	EXP Input 18 to Output 2	238Ø1
EXP Input 17 to Output 3	237Ø2	EXP Input 18 to Output 3	238Ø2
EXP Input 17 to Output 4	237Ø3	EXP Input 18 to Output 4	238Ø3
EXP Input 17 to Output 5	237Ø4	EXP Input 18 to Output 5	238Ø4
EXP Input 17 to Output 6	237Ø5	EXP Input 18 to Output 6	238Ø5
EXP Input 17 to Output 7	237Ø6	EXP Input 18 to Output 7	238Ø6
EXP Input 17 to Output 8	237Ø7	EXP Input 18 to Output 8	238Ø7
EXP Input 19 to Output 1	239ØØ	EXP Input 20 to Output 1	24ØØØ
EXP Input 19 to Output 2	239Ø1	EXP Input 20 to Output 2	24ØØ1
EXP Input 19 to Output 3	239Ø2	EXP Input 20 to Output 3	24ØØ2
EXP Input 19 to Output 4	239Ø3	EXP Input 20 to Output 4	24ØØ3
EXP Input 19 to Output 5	239Ø4	EXP Input 20 to Output 5	24ØØ4
EXP Input 19 to Output 6	239Ø5	EXP Input 20 to Output 6	24ØØ5
EXP Input 19 to Output 7	239Ø6	EXP Input 20 to Output 7	24ØØ6
EXP Input 19 to Output 8	239Ø7	EXP Input 20 to Output 8	24ØØ7

③ EXP Returns to Outputs	X60	③ EXP Returns to Outputs	X60
EXP Input 21 to Output 1	241ØØ	EXP Input 22 to Output 1	242ØØ
EXP Input 21 to Output 2	241Ø1	EXP Input 22 to Output 2	242Ø1
EXP Input 21 to Output 3	241Ø2	EXP Input 22 to Output 3	242Ø2
EXP Input 21 to Output 4	241Ø3	EXP Input 22 to Output 4	242Ø3
EXP Input 21 to Output 5	241Ø4	EXP Input 22 to Output 5	242Ø4
EXP Input 21 to Output 6	241Ø5	EXP Input 22 to Output 6	242Ø5
EXP Input 21 to Output 7	241Ø6	EXP Input 22 to Output 7	242Ø6
EXP Input 21 to Output 8	241Ø7	EXP Input 22 to Output 8	242Ø7
EXP Input 23 to Output 1	243ØØ	EXP Input 24 to Output 1	244ØØ
EXP Input 23 to Output 2	243Ø1	EXP Input 24 to Output 2	244Ø1
EXP Input 23 to Output 3	243Ø2	EXP Input 24 to Output 3	244Ø2
EXP Input 23 to Output 4	243Ø3	EXP Input 24 to Output 4	244Ø3
EXP Input 23 to Output 5	243Ø4	EXP Input 24 to Output 5	244Ø4
EXP Input 23 to Output 6	243Ø5	EXP Input 24 to Output 6	244Ø5
EXP Input 23 to Output 7	243Ø6	EXP Input 24 to Output 7	244Ø6
EXP Input 23 to Output 8	243Ø7	EXP Input 24 to Output 8	244Ø7
EXP Input 25 to Output 1	245ØØ	EXP Input 26 to Output 1	246ØØ
EXP Input 25 to Output 2	245Ø1	EXP Input 26 to Output 2	246Ø1
EXP Input 25 to Output 3	245Ø2	EXP Input 26 to Output 3	246Ø2
EXP Input 25 to Output 4	245Ø3	EXP Input 26 to Output 4	246Ø3
EXP Input 25 to Output 5	245Ø4	EXP Input 26 to Output 5	246Ø4
EXP Input 25 to Output 6	245Ø5	EXP Input 26 to Output 6	246Ø5
EXP Input 25 to Output 7	245Ø6	EXP Input 26 to Output 7	246Ø6
EXP Input 25 to Output 8	245Ø7	EXP Input 26 to Output 8	246Ø7
EXP Input 27 to Output 1	247ØØ	EXP Input 28 to Output 1	248ØØ
EXP Input 27 to Output 2	247Ø1	EXP Input 28 to Output 2	248Ø1
EXP Input 27 to Output 3	247Ø2	EXP Input 28 to Output 3	248Ø2
EXP Input 27 to Output 4	247Ø3	EXP Input 28 to Output 4	248Ø3
EXP Input 27 to Output 5	247Ø4	EXP Input 28 to Output 5	248Ø4
EXP Input 27 to Output 6	247Ø5	EXP Input 28 to Output 6	248Ø5
EXP Input 27 to Output 7	247Ø6	EXP Input 28 to Output 7	248Ø6
EXP Input 27 to Output 8	247Ø7	EXP Input 28 to Output 8	248Ø7
EXP Input 29 to Output 1	249ØØ	EXP Input 30 to Output 1	25ØØØ
EXP Input 29 to Output 2	249Ø1	EXP Input 30 to Output 2	25ØØ1
EXP Input 29 to Output 3	249Ø2	EXP Input 30 to Output 3	25ØØ2
EXP Input 29 to Output 4	249Ø3	EXP Input 30 to Output 4	25ØØ3
EXP Input 29 to Output 5	249Ø4	EXP Input 30 to Output 5	25ØØ4
EXP Input 29 to Output 6	249Ø5	EXP Input 30 to Output 6	25ØØ5
EXP Input 29 to Output 7	249Ø6	EXP Input 30 to Output 7	25ØØ6
EXP Input 29 to Output 8	249Ø7	EXP Input 30 to Output 8	25ØØ7

③ EXP Returns to Outputs	X60	③ EXP Returns to Outputs	X60
EXP Input 31 to Output 1	251ØØ	EXP Input 32 to Output 1	252ØØ
EXP Input 31 to Output 2	251Ø1	EXP Input 32 to Output 2	252Ø1
EXP Input 31 to Output 3	251Ø2	EXP Input 32 to Output 3	252Ø2
EXP Input 31 to Output 4	251Ø3	EXP Input 32 to Output 4	252Ø3
EXP Input 31 to Output 5	251Ø4	EXP Input 32 to Output 5	252Ø4
EXP Input 31 to Output 6	251Ø5	EXP Input 32 to Output 6	252Ø5
EXP Input 31 to Output 7	251Ø6	EXP Input 32 to Output 7	252Ø6
EXP Input 31 to Output 8	251Ø7	EXP Input 32 to Output 8	252Ø7
EXP Input 33 to Output 1	253ØØ	EXP Input 34 to Output 1	254ØØ
EXP Input 33 to Output 2	253Ø1	EXP Input 34 to Output 2	254Ø1
EXP Input 33 to Output 3	253Ø2	EXP Input 34 to Output 3	254Ø2
EXP Input 33 to Output 4	253Ø3	EXP Input 34 to Output 4	254Ø3
EXP Input 33 to Output 5	253Ø4	EXP Input 34 to Output 5	254Ø4
EXP Input 33 to Output 6	253Ø5	EXP Input 34 to Output 6	254Ø5
EXP Input 33 to Output 7	253Ø6	EXP Input 34 to Output 7	254Ø6
EXP Input 33 to Output 8	253Ø7	EXP Input 34 to Output 8	254Ø7
EXP Input 35 to Output 1	255ØØ	EXP Input 36 to Output 1	256ØØ
EXP Input 35 to Output 2	255Ø1	EXP Input 36 to Output 2	256Ø1
EXP Input 35 to Output 3	255Ø2	EXP Input 36 to Output 3	256Ø2
EXP Input 35 to Output 4	255Ø3	EXP Input 36 to Output 4	256Ø3
EXP Input 35 to Output 5	255Ø4	EXP Input 36 to Output 5	256Ø4
EXP Input 35 to Output 6	255Ø5	EXP Input 36 to Output 6	256Ø5
EXP Input 35 to Output 7	255Ø6	EXP Input 36 to Output 7	256Ø6
EXP Input 35 to Output 8	255Ø7	EXP Input 36 to Output 8	256Ø7
EXP Input 37 to Output 1	257ØØ	EXP Input 38 to Output 1	258ØØ
EXP Input 37 to Output 2	257Ø1	EXP Input 38 to Output 2	258Ø1
EXP Input 37 to Output 3	257Ø2	EXP Input 38 to Output 3	258Ø2
EXP Input 37 to Output 4	257Ø3	EXP Input 38 to Output 4	258Ø3
EXP Input 37 to Output 5	257Ø4	EXP Input 38 to Output 5	258Ø4
EXP Input 37 to Output 6	257Ø5	EXP Input 38 to Output 6	258Ø5
EXP Input 37 to Output 7	257Ø6	EXP Input 38 to Output 7	258Ø6
EXP Input 37 to Output 8	257Ø7	EXP Input 38 to Output 8	258Ø7
EXP Input 39 to Output 1	259ØØ	EXP Input 40 to Output 1	26ØØØ
EXP Input 39 to Output 2	259Ø1	EXP Input 40 to Output 2	26ØØ1
EXP Input 39 to Output 3	259Ø2	EXP Input 40 to Output 3	26ØØ2
EXP Input 39 to Output 4	259Ø3	EXP Input 40 to Output 4	26ØØ3
EXP Input 39 to Output 5	259Ø4	EXP Input 40 to Output 5	26ØØ4
EXP Input 39 to Output 6	259Ø5	EXP Input 40 to Output 6	26ØØ5
EXP Input 39 to Output 7	259Ø6	EXP Input 40 to Output 7	26ØØ6
EXP Input 39 to Output 8	259Ø7	EXP Input 40 to Output 8	26ØØ7

③ EXP Returns to Outputs	X60	③ EXP Returns to Outputs	X60
EXP Input 41 to Output 1	261ØØ	EXP Input 42 to Output 1	262ØØ
EXP Input 41 to Output 2	261Ø1	EXP Input 42 to Output 2	262Ø1
EXP Input 41 to Output 3	261Ø2	EXP Input 42 to Output 3	262Ø2
EXP Input 41 to Output 4	261Ø3	EXP Input 42 to Output 4	262Ø3
EXP Input 41 to Output 5	261Ø4	EXP Input 42 to Output 5	262Ø4
EXP Input 41 to Output 6	261Ø5	EXP Input 42 to Output 6	262Ø5
EXP Input 41 to Output 7	261Ø6	EXP Input 42 to Output 7	262Ø6
EXP Input 41 to Output 8	261Ø7	EXP Input 42 to Output 8	262Ø7
EXP Input 43 to Output 1	263ØØ	EXP Input 44 to Output 1	264ØØ
EXP Input 43 to Output 2	263Ø1	EXP Input 44 to Output 2	264Ø1
EXP Input 43 to Output 3	263Ø2	EXP Input 44 to Output 3	264Ø2
EXP Input 43 to Output 4	263Ø3	EXP Input 44 to Output 4	264Ø3
EXP Input 43 to Output 5	263Ø4	EXP Input 44 to Output 5	264Ø4
EXP Input 43 to Output 6	263Ø5	EXP Input 44 to Output 6	264Ø5
EXP Input 43 to Output 7	263Ø6	EXP Input 44 to Output 7	264Ø6
EXP Input 43 to Output 8	263Ø7	EXP Input 44 to Output 8	264Ø7
EXP Input 45 to Output 1	265ØØ	EXP Input 46 to Output 1	266ØØ
EXP Input 45 to Output 2	265Ø1	EXP Input 46 to Output 2	266Ø1
EXP Input 45 to Output 3	265Ø2	EXP Input 46 to Output 3	266Ø2
EXP Input 45 to Output 4	265Ø3	EXP Input 46 to Output 4	266Ø3
EXP Input 45 to Output 5	265Ø4	EXP Input 46 to Output 5	266Ø4
EXP Input 45 to Output 6	265Ø5	EXP Input 46 to Output 6	266Ø5
EXP Input 45 to Output 7	265Ø6	EXP Input 46 to Output 7	266Ø6
EXP Input 45 to Output 8	265Ø7	EXP Input 46 to Output 8	266Ø7
EXP Input 47 to Output 1	267ØØ	EXP Input 48 to Output 1	268ØØ
EXP Input 47 to Output 2	267Ø1	EXP Input 48 to Output 2	268Ø1
EXP Input 47 to Output 3	267Ø2	EXP Input 48 to Output 3	268Ø2
EXP Input 47 to Output 4	267Ø3	EXP Input 48 to Output 4	268Ø3
EXP Input 47 to Output 5	267Ø4	EXP Input 48 to Output 5	268Ø4
EXP Input 47 to Output 6	267Ø5	EXP Input 48 to Output 6	268Ø5
EXP Input 47 to Output 7	267Ø6	EXP Input 48 to Output 7	268Ø6
EXP Input 47 to Output 8	267Ø7	EXP Input 48 to Output 8	268Ø7
EXP Input 49 to Output 1	269ØØ	EXP Input 50 to Output 1	27ØØØ
EXP Input 49 to Output 2	269Ø1	EXP Input 50 to Output 2	27ØØ1
EXP Input 49 to Output 3	269Ø2	EXP Input 50 to Output 3	27ØØ2
EXP Input 49 to Output 4	269Ø3	EXP Input 50 to Output 4	27ØØ3
EXP Input 49 to Output 5	269Ø4	EXP Input 50 to Output 5	27ØØ4
EXP Input 49 to Output 6	269Ø5	EXP Input 50 to Output 6	27ØØ5
EXP Input 49 to Output 7	269Ø6	EXP Input 50 to Output 7	27ØØ6
EXP Input 49 to Output 8	269Ø7	EXP Input 50 to Output 8	27ØØ7

③ EXP Returns to Outputs	X60	③ EXP Returns to Outputs	X60
EXP Input 51 to Output 1	271ØØ	EXP Input 52 to Output 1	272ØØ
EXP Input 51 to Output 2	271Ø1	EXP Input 52 to Output 2	272Ø1
EXP Input 51 to Output 3	271Ø2	EXP Input 52 to Output 3	272Ø2
EXP Input 51 to Output 4	271Ø3	EXP Input 52 to Output 4	272Ø3
EXP Input 51 to Output 5	271Ø4	EXP Input 52 to Output 5	272Ø4
EXP Input 51 to Output 6	271Ø5	EXP Input 52 to Output 6	272Ø5
EXP Input 51 to Output 7	271Ø6	EXP Input 52 to Output 7	272Ø6
EXP Input 51 to Output 8	271Ø7	EXP Input 52 to Output 8	272Ø7
EXP Input 53 to Output 1	273ØØ	EXP Input 54 to Output 1	274ØØ
EXP Input 53 to Output 2	273Ø1	EXP Input 54 to Output 2	274Ø1
EXP Input 53 to Output 3	273Ø2	EXP Input 54 to Output 3	274Ø2
EXP Input 53 to Output 4	273Ø3	EXP Input 54 to Output 4	274Ø3
EXP Input 53 to Output 5	273Ø4	EXP Input 54 to Output 5	274Ø4
EXP Input 53 to Output 6	273Ø5	EXP Input 54 to Output 6	274Ø5
EXP Input 53 to Output 7	273Ø6	EXP Input 54 to Output 7	274Ø6
EXP Input 53 to Output 8	273Ø7	EXP Input 54 to Output 8	274Ø7
EXP Input 55 to Output 1	275ØØ	EXP Input 56 to Output 1	276ØØ
EXP Input 55 to Output 2	275Ø1	EXP Input 56 to Output 2	276Ø1
EXP Input 55 to Output 3	275Ø2	EXP Input 56 to Output 3	276Ø2
EXP Input 55 to Output 4	275Ø3	EXP Input 56 to Output 4	276Ø3
EXP Input 55 to Output 5	275Ø4	EXP Input 56 to Output 5	276Ø4
EXP Input 55 to Output 6	275Ø5	EXP Input 56 to Output 6	276Ø5
EXP Input 55 to Output 7	275Ø6	EXP Input 56 to Output 7	276Ø6
EXP Input 55 to Output 8	275Ø7	EXP Input 56 to Output 8	276Ø7

Virtual Sends Bus Mix Matrix

The Virtual Sends bus mixer can be further broken down into the mic/line inputs (①), virtual bus returns (②), and EXP input mix-points (③).

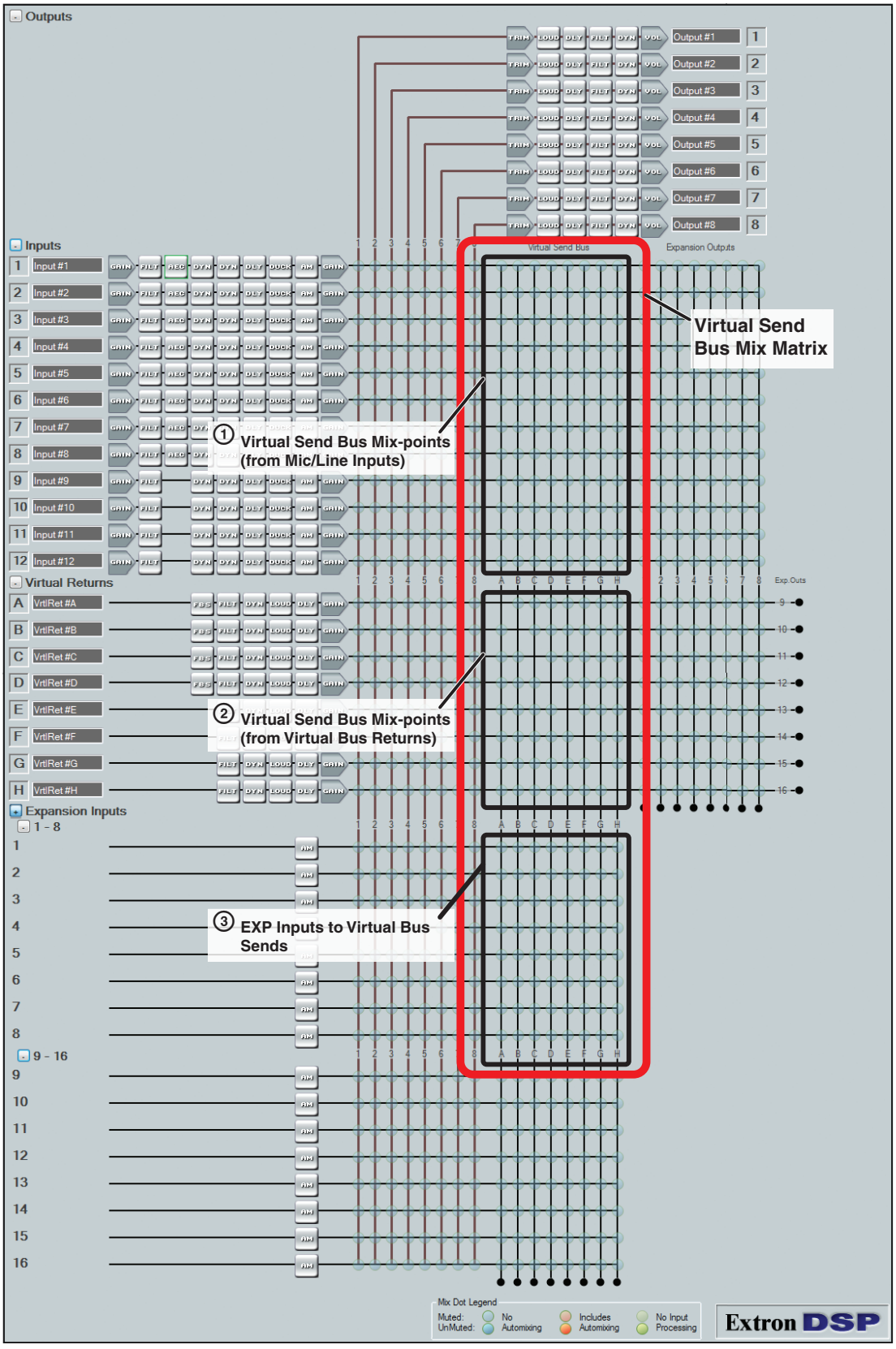


Table 6: Mic/Line Inputs to Virtual Sends Mix-points

① Inputs to Virtual Sends	X60	① Inputs to Virtual Sends	X60
Input 1 to Virtual Send A	2ØØØ9	Input 2 to Virtual Send A	2Ø1Ø9
Input 1 to Virtual Send B	2ØØ1Ø	Input 2 to Virtual Send B	2Ø11Ø
Input 1 to Virtual Send C	2ØØ11	Input 2 to Virtual Send C	2Ø111
Input 1 to Virtual Send D	2ØØ12	Input 2 to Virtual Send D	2Ø112
Input 1 to Virtual Send E	2ØØ13	Input 2 to Virtual Send E	2Ø113
Input 1 to Virtual Send F	2ØØ14	Input 2 to Virtual Send F	2Ø114
Input 1 to Virtual Send G	2ØØ15	Input 2 to Virtual Send G	2Ø115
Input 1 to Virtual Send H	2ØØ16	Input 2 to Virtual Send H	2Ø116
Input 3 to Virtual Send A	2Ø2Ø9	Input 4 to Virtual Send A	2Ø3Ø9
Input 3 to Virtual Send B	2Ø21Ø	Input 4 to Virtual Send B	2Ø31Ø
Input 3 to Virtual Send C	2Ø211	Input 4 to Virtual Send C	2Ø311
Input 3 to Virtual Send D	2Ø212	Input 4 to Virtual Send D	2Ø312
Input 3 to Virtual Send E	2Ø213	Input 4 to Virtual Send E	2Ø313
Input 3 to Virtual Send F	2Ø214	Input 4 to Virtual Send F	2Ø314
Input 3 to Virtual Send G	2Ø215	Input 4 to Virtual Send G	2Ø315
Input 3 to Virtual Send H	2Ø216	Input 4 to Virtual Send H	2Ø316
Input 5 to Virtual Send A	2Ø4Ø9	Input 6 to Virtual Send A	2Ø5Ø9
Input 5 to Virtual Send B	2Ø41Ø	Input 6 to Virtual Send B	2Ø51Ø
Input 5 to Virtual Send C	2Ø411	Input 6 to Virtual Send C	2Ø511
Input 5 to Virtual Send D	2Ø412	Input 6 to Virtual Send D	2Ø512
Input 5 to Virtual Send E	2Ø413	Input 6 to Virtual Send E	2Ø513
Input 5 to Virtual Send F	2Ø414	Input 6 to Virtual Send F	2Ø514
Input 5 to Virtual Send G	2Ø415	Input 6 to Virtual Send G	2Ø515
Input 5 to Virtual Send H	2Ø416	Input 6 to Virtual Send H	2Ø516
Input 7 to Virtual Send A	2Ø6Ø9	Input 8 to Virtual Send A	2Ø7Ø9
Input 7 to Virtual Send B	2Ø61Ø	Input 8 to Virtual Send B	2Ø71Ø
Input 7 to Virtual Send C	2Ø611	Input 8 to Virtual Send C	2Ø711
Input 7 to Virtual Send D	2Ø612	Input 8 to Virtual Send D	2Ø712
Input 7 to Virtual Send E	2Ø613	Input 8 to Virtual Send E	2Ø713
Input 7 to Virtual Send F	2Ø614	Input 8 to Virtual Send F	2Ø714
Input 7 to Virtual Send G	2Ø615	Input 8 to Virtual Send G	2Ø715
Input 7 to Virtual Send H	2Ø616	Input 8 to Virtual Send H	2Ø716
Input 9 to Virtual Send A	2Ø8Ø9	Input 10 to Virtual Send A	2Ø9Ø9
Input 9 to Virtual Send B	2Ø81Ø	Input 10 to Virtual Send B	2Ø91Ø
Input 9 to Virtual Send C	2Ø811	Input 10 to Virtual Send C	2Ø911
Input 9 to Virtual Send D	2Ø812	Input 10 to Virtual Send D	2Ø912
Input 9 to Virtual Send E	2Ø813	Input 10 to Virtual Send E	2Ø913
Input 9 to Virtual Send F	2Ø814	Input 10 to Virtual Send F	2Ø914
Input 9 to Virtual Send G	2Ø815	Input 10 to Virtual Send G	2Ø915
Input 9 to Virtual Send H	2Ø816	Input 10 to Virtual Send H	2Ø916

① Inputs to Virtual Sends	X60	① Inputs to Virtual Sends	X60
Input 11 to Virtual Send A	21ØØ9	Input 12 to Virtual Send A	211Ø9
Input 11 to Virtual Send B	21Ø1Ø	Input 12 to Virtual Send B	2111Ø
Input 11 to Virtual Send C	21Ø11	Input 12 to Virtual Send C	21111
Input 11 to Virtual Send D	21Ø12	Input 12 to Virtual Send D	21112
Input 11 to Virtual Send E	21Ø13	Input 12 to Virtual Send E	21113
Input 11 to Virtual Send F	21Ø14	Input 12 to Virtual Send F	21114
Input 11 to Virtual Send G	21Ø15	Input 12 to Virtual Send G	21115
Input 11 to Virtual Send H	21Ø16	Input 12 to Virtual Send H	21116

Telephone Rx to Virtual Send A	212Ø9
Telephone Rx to Virtual Send B	2121Ø
Telephone Rx to Virtual Send C	21211
Telephone Rx to Virtual Send D	21212
Telephone Rx to Virtual Send E	21213
Telephone Rx to Virtual Send F	21214
Telephone Rx to Virtual Send G	21215
Telephone Rx to Virtual Send H	21216

Table 7: Virtual Returns to Virtual Sends

② Virtual Returns to Virtual Sends	X60	② Virtual Returns to Virtual Sends	X60
		Virtual Return B to Virtual Send A	214Ø9
Virtual Return A to Virtual Send B	2131Ø		
Virtual Return A to Virtual Send C	21311	Virtual Return B to Virtual Send C	21411
Virtual Return A to Virtual Send D	21312	Virtual Return B to Virtual Send D	21412
Virtual Return A to Virtual Send E	21313	Virtual Return B to Virtual Send E	21413
Virtual Return A to Virtual Send F	21314	Virtual Return B to Virtual Send F	21414
Virtual Return A to Virtual Send G	21315	Virtual Return B to Virtual Send G	21415
Virtual Return A to Virtual Send H	21316	Virtual Return B to Virtual Send H	21416
Virtual Return C to Virtual Send A	215Ø9	Virtual Return D to Virtual Send A	216Ø9
Virtual Return C to Virtual Send B	2151Ø	Virtual Return D to Virtual Send B	2161Ø
		Virtual Return D to Virtual Send C	21611
Virtual Return C to Virtual Send D	21512		
Virtual Return C to Virtual Send E	21513	Virtual Return D to Virtual Send E	21613
Virtual Return C to Virtual Send F	21514	Virtual Return D to Virtual Send F	21614
Virtual Return C to Virtual Send G	21515	Virtual Return D to Virtual Send G	21615
Virtual Return C to Virtual Send H	21516	Virtual Return D to Virtual Send H	21616
Virtual Return E to Virtual Send A	217Ø9	Virtual Return F to Virtual Send A	218Ø9
Virtual Return E to Virtual Send B	2171Ø	Virtual Return F to Virtual Send B	2181Ø
Virtual Return E to Virtual Send C	21711	Virtual Return F to Virtual Send C	21811
Virtual Return E to Virtual Send D	21712	Virtual Return F to Virtual Send D	21812
		Virtual Return F to Virtual Send E	21813
Virtual Return E to Virtual Send F	21714		
Virtual Return E to Virtual Send G	21715	Virtual Return F to Virtual Send G	21815
Virtual Return E to Virtual Send H	21716	Virtual Return F to Virtual Send H	21816
Virtual Return G to Virtual Send A	219Ø9	Virtual Return H to Virtual Send A	22ØØ9
Virtual Return G to Virtual Send B	2191Ø	Virtual Return H to Virtual Send B	22Ø1Ø
Virtual Return G to Virtual Send C	21911	Virtual Return H to Virtual Send C	22Ø11
Virtual Return G to Virtual Send D	21912	Virtual Return H to Virtual Send D	22Ø12
Virtual Return G to Virtual Send E	21913	Virtual Return H to Virtual Send E	22Ø13
Virtual Return G to Virtual Send F	21914	Virtual Return H to Virtual Send F	22Ø14
		Virtual Return H to Virtual Send G	22Ø15
Virtual Return G to Virtual Send H	21916		

Table 8: EXP Bus Inputs to Virtual Sends

③ EXP Inputs to Virtual Sends	X60	③ EXP Inputs to Virtual Sends	X60
EXP Input 1 to Virtual Sends A	221Ø9	EXP Input 2 to Virtual Sends A	222Ø9
EXP Input 1 to Virtual Sends B	2211Ø	EXP Input 2 to Virtual Sends B	2221Ø
EXP Input 1 to Virtual Sends C	22111	EXP Input 2 to Virtual Sends C	22211
EXP Input 1 to Virtual Sends D	22112	EXP Input 2 to Virtual Sends D	22212
EXP Input 1 to Virtual Sends E	22113	EXP Input 2 to Virtual Sends E	22213
EXP Input 1 to Virtual Sends F	22114	EXP Input 2 to Virtual Sends F	22214
EXP Input 1 to Virtual Sends G	22115	EXP Input 2 to Virtual Sends G	22215
EXP Input 1 to Virtual Sends H	22116	EXP Input 2 to Virtual Sends H	22216
EXP Input 3 to Virtual Sends A	223Ø9	EXP Input 4 to Virtual Sends A	224Ø9
EXP Input 3 to Virtual Sends B	2231Ø	EXP Input 4 to Virtual Sends B	2241Ø
EXP Input 3 to Virtual Sends C	22311	EXP Input 4 to Virtual Sends C	22411
EXP Input 3 to Virtual Sends D	22312	EXP Input 4 to Virtual Sends D	22412
EXP Input 3 to Virtual Sends E	22313	EXP Input 4 to Virtual Sends E	22413
EXP Input 3 to Virtual Sends F	22314	EXP Input 4 to Virtual Sends F	22414
EXP Input 3 to Virtual Sends G	22315	EXP Input 4 to Virtual Sends G	22415
EXP Input 3 to Virtual Sends H	22316	EXP Input 4 to Virtual Sends H	22416
EXP Input 5 to Virtual Sends A	225Ø9	EXP Input 6 to Virtual Sends A	226Ø9
EXP Input 5 to Virtual Sends B	2251Ø	EXP Input 6 to Virtual Sends B	2261Ø
EXP Input 5 to Virtual Sends C	22511	EXP Input 6 to Virtual Sends C	22611
EXP Input 5 to Virtual Sends D	22512	EXP Input 6 to Virtual Sends D	22612
EXP Input 5 to Virtual Sends E	22513	EXP Input 6 to Virtual Sends E	22613
EXP Input 5 to Virtual Sends F	22514	EXP Input 6 to Virtual Sends F	22614
EXP Input 5 to Virtual Sends G	22515	EXP Input 6 to Virtual Sends G	22615
EXP Input 5 to Virtual Sends H	22516	EXP Input 6 to Virtual Sends H	22616
EXP Input 7 to Virtual Sends A	227Ø9	EXP Input 8 to Virtual Sends A	228Ø9
EXP Input 7 to Virtual Sends B	2271Ø	EXP Input 8 to Virtual Sends B	2281Ø
EXP Input 7 to Virtual Sends C	22711	EXP Input 8 to Virtual Sends C	22811
EXP Input 7 to Virtual Sends D	22712	EXP Input 8 to Virtual Sends D	22812
EXP Input 7 to Virtual Sends E	22713	EXP Input 8 to Virtual Sends E	22813
EXP Input 7 to Virtual Sends F	22714	EXP Input 8 to Virtual Sends F	22814
EXP Input 7 to Virtual Sends G	22715	EXP Input 8 to Virtual Sends G	22815
EXP Input 7 to Virtual Sends H	22716	EXP Input 8 to Virtual Sends H	22816
EXP Input 9 to Virtual Sends A	229Ø9	EXP Input 10 to Virtual Sends A	23ØØ9
EXP Input 9 to Virtual Sends B	2291Ø	EXP Input 10 to Virtual Sends B	23Ø1Ø
EXP Input 9 to Virtual Sends C	22911	EXP Input 10 to Virtual Sends C	23Ø11
EXP Input 9 to Virtual Sends D	22912	EXP Input 10 to Virtual Sends D	23Ø12
EXP Input 9 to Virtual Sends E	22913	EXP Input 10 to Virtual Sends E	23Ø13
EXP Input 9 to Virtual Sends F	22914	EXP Input 10 to Virtual Sends F	23Ø14
EXP Input 9 to Virtual Sends G	22915	EXP Input 10 to Virtual Sends G	23Ø15
EXP Input 9 to Virtual Sends H	22916	EXP Input 10 to Virtual Sends H	23Ø16

③ EXP Inputs to Virtual Sends	X60	③ EXP Inputs to Virtual Sends	X60
EXP Input 11 to Virtual Sends A	231Ø9	EXP Input 12 to Virtual Sends A	232Ø9
EXP Input 11 to Virtual Sends B	2311Ø	EXP Input 12 to Virtual Sends B	2321Ø
EXP Input 11 to Virtual Sends C	23111	EXP Input 12 to Virtual Sends C	23211
EXP Input 11 to Virtual Sends D	23112	EXP Input 12 to Virtual Sends D	23212
EXP Input 11 to Virtual Sends E	23113	EXP Input 12 to Virtual Sends E	23213
EXP Input 11 to Virtual Sends F	23114	EXP Input 12 to Virtual Sends F	23214
EXP Input 11 to Virtual Sends G	23115	EXP Input 12 to Virtual Sends G	23215
EXP Input 11 to Virtual Sends H	23116	EXP Input 12 to Virtual Sends H	23216
EXP Input 13 to Virtual Sends A	233Ø9	EXP Input 14 to Virtual Sends A	234Ø9
EXP Input 13 to Virtual Sends B	2331Ø	EXP Input 14 to Virtual Sends B	2341Ø
EXP Input 13 to Virtual Sends C	23311	EXP Input 14 to Virtual Sends C	23411
EXP Input 13 to Virtual Sends D	23312	EXP Input 14 to Virtual Sends D	23412
EXP Input 13 to Virtual Sends E	23313	EXP Input 14 to Virtual Sends E	23413
EXP Input 13 to Virtual Sends F	23314	EXP Input 14 to Virtual Sends F	23414
EXP Input 13 to Virtual Sends G	23315	EXP Input 14 to Virtual Sends G	23415
EXP Input 13 to Virtual Sends H	23316	EXP Input 14 to Virtual Sends H	23416
EXP Input 15 to Virtual Sends A	235Ø9	EXP Input 16 to Virtual Sends A	236Ø9
EXP Input 15 to Virtual Sends B	2351Ø	EXP Input 16 to Virtual Sends B	2361Ø
EXP Input 15 to Virtual Sends C	23511	EXP Input 16 to Virtual Sends C	23611
EXP Input 15 to Virtual Sends D	23512	EXP Input 16 to Virtual Sends D	23612
EXP Input 15 to Virtual Sends E	23513	EXP Input 16 to Virtual Sends E	23613
EXP Input 15 to Virtual Sends F	23514	EXP Input 16 to Virtual Sends F	23614
EXP Input 15 to Virtual Sends G	23515	EXP Input 16 to Virtual Sends G	23615
EXP Input 15 to Virtual Sends H	23516	EXP Input 16 to Virtual Sends H	23616
EXP Input 17 to Virtual Sends A	237Ø9	EXP Input 18 to Virtual Sends A	238Ø9
EXP Input 17 to Virtual Sends B	2371Ø	EXP Input 18 to Virtual Sends B	2381Ø
EXP Input 17 to Virtual Sends C	23711	EXP Input 18 to Virtual Sends C	23811
EXP Input 17 to Virtual Sends D	23712	EXP Input 18 to Virtual Sends D	23812
EXP Input 17 to Virtual Sends E	23713	EXP Input 18 to Virtual Sends E	23813
EXP Input 17 to Virtual Sends F	23714	EXP Input 18 to Virtual Sends F	23814
EXP Input 17 to Virtual Sends G	23715	EXP Input 18 to Virtual Sends G	23815
EXP Input 17 to Virtual Sends H	23716	EXP Input 18 to Virtual Sends H	23816
EXP Input 19 to Virtual Sends A	239Ø9	EXP Input 20 to Virtual Sends A	24ØØ9
EXP Input 19 to Virtual Sends B	2391Ø	EXP Input 20 to Virtual Sends B	24Ø1Ø
EXP Input 19 to Virtual Sends C	23911	EXP Input 20 to Virtual Sends C	24Ø11
EXP Input 19 to Virtual Sends D	23912	EXP Input 20 to Virtual Sends D	24Ø12
EXP Input 19 to Virtual Sends E	23913	EXP Input 20 to Virtual Sends E	24Ø13
EXP Input 19 to Virtual Sends F	23914	EXP Input 20 to Virtual Sends F	24Ø14
EXP Input 19 to Virtual Sends G	23915	EXP Input 20 to Virtual Sends G	24Ø15
EXP Input 19 to Virtual Sends H	23916	EXP Input 20 to Virtual Sends H	24Ø16

③ EXP Inputs to Virtual Sends	X60	③ EXP Inputs to Virtual Sends	X60
EXP Input 21 to Virtual Sends A	231Ø9	EXP Input 22 to Virtual Sends A	232Ø9
EXP Input 21 to Virtual Sends B	2311Ø	EXP Input 22 to Virtual Sends B	2321Ø
EXP Input 21 to Virtual Sends C	23111	EXP Input 22 to Virtual Sends C	23211
EXP Input 21 to Virtual Sends D	23112	EXP Input 22 to Virtual Sends D	23212
EXP Input 21 to Virtual Sends E	23113	EXP Input 22 to Virtual Sends E	23213
EXP Input 21 to Virtual Sends F	23114	EXP Input 22 to Virtual Sends F	23214
EXP Input 21 to Virtual Sends G	23115	EXP Input 22 to Virtual Sends G	23215
EXP Input 21 to Virtual Sends H	23116	EXP Input 22 to Virtual Sends H	23216
EXP Input 23 to Virtual Sends A	233Ø9	EXP Input 24 to Virtual Sends A	234Ø9
EXP Input 23 to Virtual Sends B	2331Ø	EXP Input 24 to Virtual Sends B	2341Ø
EXP Input 23 to Virtual Sends C	23311	EXP Input 24 to Virtual Sends C	23411
EXP Input 23 to Virtual Sends D	23312	EXP Input 24 to Virtual Sends D	23412
EXP Input 23 to Virtual Sends E	23313	EXP Input 24 to Virtual Sends E	23413
EXP Input 23 to Virtual Sends F	23314	EXP Input 24 to Virtual Sends F	23414
EXP Input 23 to Virtual Sends G	23315	EXP Input 24 to Virtual Sends G	23415
EXP Input 23 to Virtual Sends H	23316	EXP Input 24 to Virtual Sends H	23416
EXP Input 25 to Virtual Sends A	235Ø9	EXP Input 26 to Virtual Sends A	236Ø9
EXP Input 25 to Virtual Sends B	2351Ø	EXP Input 26 to Virtual Sends B	2361Ø
EXP Input 25 to Virtual Sends C	23511	EXP Input 26 to Virtual Sends C	23611
EXP Input 25 to Virtual Sends D	23512	EXP Input 26 to Virtual Sends D	23612
EXP Input 25 to Virtual Sends E	23513	EXP Input 26 to Virtual Sends E	23613
EXP Input 25 to Virtual Sends F	23514	EXP Input 26 to Virtual Sends F	23614
EXP Input 25 to Virtual Sends G	23515	EXP Input 26 to Virtual Sends G	23615
EXP Input 25 to Virtual Sends H	23516	EXP Input 26 to Virtual Sends H	23616
EXP Input 27 to Virtual Sends A	237Ø9	EXP Input 28 to Virtual Sends A	238Ø9
EXP Input 27 to Virtual Sends B	2371Ø	EXP Input 28 to Virtual Sends B	2381Ø
EXP Input 27 to Virtual Sends C	23711	EXP Input 28 to Virtual Sends C	23811
EXP Input 27 to Virtual Sends D	23712	EXP Input 28 to Virtual Sends D	23812
EXP Input 27 to Virtual Sends E	23713	EXP Input 28 to Virtual Sends E	23813
EXP Input 27 to Virtual Sends F	23714	EXP Input 28 to Virtual Sends F	23814
EXP Input 27 to Virtual Sends G	23715	EXP Input 28 to Virtual Sends G	23815
EXP Input 27 to Virtual Sends H	23716	EXP Input 28 to Virtual Sends H	23816
EXP Input 29 to Virtual Sends A	239Ø9	EXP Input 30 to Virtual Sends A	24ØØ9
EXP Input 29 to Virtual Sends B	2391Ø	EXP Input 30 to Virtual Sends B	24Ø1Ø
EXP Input 29 to Virtual Sends C	23911	EXP Input 30 to Virtual Sends C	24Ø11
EXP Input 29 to Virtual Sends D	23912	EXP Input 30 to Virtual Sends D	24Ø12
EXP Input 29 to Virtual Sends E	23913	EXP Input 30 to Virtual Sends E	24Ø13
EXP Input 29 to Virtual Sends F	23914	EXP Input 30 to Virtual Sends F	24Ø14
EXP Input 29 to Virtual Sends G	23915	EXP Input 30 to Virtual Sends G	24Ø15
EXP Input 29 to Virtual Sends H	23916	EXP Input 30 to Virtual Sends H	24Ø16

③ EXP Inputs to Virtual Sends	X60	③ EXP Inputs to Virtual Sends	X60
EXP Input 31 to Virtual Sends A	241Ø9	EXP Input 32 to Virtual Sends A	242Ø9
EXP Input 31 to Virtual Sends B	2411Ø	EXP Input 32 to Virtual Sends B	2421Ø
EXP Input 31 to Virtual Sends C	24111	EXP Input 32 to Virtual Sends C	24211
EXP Input 31 to Virtual Sends D	24112	EXP Input 32 to Virtual Sends D	24212
EXP Input 31 to Virtual Sends E	24113	EXP Input 32 to Virtual Sends E	24213
EXP Input 31 to Virtual Sends F	24114	EXP Input 32 to Virtual Sends F	24214
EXP Input 31 to Virtual Sends G	24115	EXP Input 32 to Virtual Sends G	24215
EXP Input 31 to Virtual Sends H	24116	EXP Input 32 to Virtual Sends H	24216
EXP Input 33 to Virtual Sends A	243Ø9	EXP Input 34 to Virtual Sends A	244Ø9
EXP Input 33 to Virtual Sends B	2431Ø	EXP Input 34 to Virtual Sends B	2441Ø
EXP Input 33 to Virtual Sends C	24311	EXP Input 34 to Virtual Sends C	24411
EXP Input 33 to Virtual Sends D	24312	EXP Input 34 to Virtual Sends D	24412
EXP Input 33 to Virtual Sends E	24313	EXP Input 34 to Virtual Sends E	24413
EXP Input 33 to Virtual Sends F	24314	EXP Input 34 to Virtual Sends F	24414
EXP Input 33 to Virtual Sends G	24315	EXP Input 34 to Virtual Sends G	24415
EXP Input 33 to Virtual Sends H	24316	EXP Input 34 to Virtual Sends H	24416
EXP Input 35 to Virtual Sends A	245Ø9	EXP Input 36 to Virtual Sends A	246Ø9
EXP Input 35 to Virtual Sends B	2451Ø	EXP Input 36 to Virtual Sends B	2461Ø
EXP Input 35 to Virtual Sends C	24511	EXP Input 36 to Virtual Sends C	24611
EXP Input 35 to Virtual Sends D	24512	EXP Input 36 to Virtual Sends D	24612
EXP Input 35 to Virtual Sends E	24513	EXP Input 36 to Virtual Sends E	24613
EXP Input 35 to Virtual Sends F	24514	EXP Input 36 to Virtual Sends F	24614
EXP Input 35 to Virtual Sends G	24515	EXP Input 36 to Virtual Sends G	24615
EXP Input 35 to Virtual Sends H	24516	EXP Input 36 to Virtual Sends H	24616
EXP Input 37 to Virtual Sends A	247Ø9	EXP Input 38 to Virtual Sends A	248Ø9
EXP Input 37 to Virtual Sends B	2471Ø	EXP Input 38 to Virtual Sends B	2481Ø
EXP Input 37 to Virtual Sends C	24711	EXP Input 38 to Virtual Sends C	24811
EXP Input 37 to Virtual Sends D	24712	EXP Input 38 to Virtual Sends D	24812
EXP Input 37 to Virtual Sends E	24713	EXP Input 38 to Virtual Sends E	24813
EXP Input 37 to Virtual Sends F	24714	EXP Input 38 to Virtual Sends F	24814
EXP Input 37 to Virtual Sends G	24715	EXP Input 38 to Virtual Sends G	24815
EXP Input 37 to Virtual Sends H	24716	EXP Input 38 to Virtual Sends H	24816
EXP Input 39 to Virtual Sends A	249Ø9	EXP Input 40 to Virtual Sends A	25ØØ9
EXP Input 39 to Virtual Sends B	2491Ø	EXP Input 40 to Virtual Sends B	25Ø1Ø
EXP Input 39 to Virtual Sends C	24911	EXP Input 40 to Virtual Sends C	25Ø11
EXP Input 39 to Virtual Sends D	24912	EXP Input 40 to Virtual Sends D	25Ø12
EXP Input 39 to Virtual Sends E	24913	EXP Input 40 to Virtual Sends E	25Ø13
EXP Input 39 to Virtual Sends F	24914	EXP Input 40 to Virtual Sends F	25Ø14
EXP Input 39 to Virtual Sends G	24915	EXP Input 40 to Virtual Sends G	25Ø15
EXP Input 39 to Virtual Sends H	24916	EXP Input 40 to Virtual Sends H	25Ø16

③ EXP Inputs to Virtual Sends	X60	③ EXP Inputs to Virtual Sends	X60
EXP Input 41 to Virtual Sends A	251Ø9	EXP Input 42 to Virtual Sends A	252Ø9
EXP Input 41 to Virtual Sends B	2511Ø	EXP Input 42 to Virtual Sends B	2521Ø
EXP Input 41 to Virtual Sends C	25111	EXP Input 42 to Virtual Sends C	25211
EXP Input 41 to Virtual Sends D	25112	EXP Input 42 to Virtual Sends D	25212
EXP Input 41 to Virtual Sends E	25113	EXP Input 42 to Virtual Sends E	25213
EXP Input 41 to Virtual Sends F	25114	EXP Input 42 to Virtual Sends F	25214
EXP Input 41 to Virtual Sends G	25115	EXP Input 42 to Virtual Sends G	25215
EXP Input 41 to Virtual Sends H	25116	EXP Input 42 to Virtual Sends H	25216
EXP Input 43 to Virtual Sends A	253Ø9	EXP Input 44 to Virtual Sends A	254Ø9
EXP Input 43 to Virtual Sends B	2531Ø	EXP Input 44 to Virtual Sends B	2541Ø
EXP Input 43 to Virtual Sends C	25311	EXP Input 44 to Virtual Sends C	25411
EXP Input 43 to Virtual Sends D	25312	EXP Input 44 to Virtual Sends D	25412
EXP Input 43 to Virtual Sends E	25313	EXP Input 44 to Virtual Sends E	25413
EXP Input 43 to Virtual Sends F	25314	EXP Input 44 to Virtual Sends F	25414
EXP Input 43 to Virtual Sends G	25315	EXP Input 44 to Virtual Sends G	25415
EXP Input 43 to Virtual Sends H	25316	EXP Input 44 to Virtual Sends H	25416
EXP Input 45 to Virtual Sends A	255Ø9	EXP Input 46 to Virtual Sends A	256Ø9
EXP Input 45 to Virtual Sends B	2551Ø	EXP Input 46 to Virtual Sends B	2561Ø
EXP Input 45 to Virtual Sends C	25511	EXP Input 46 to Virtual Sends C	25611
EXP Input 45 to Virtual Sends D	25512	EXP Input 46 to Virtual Sends D	25612
EXP Input 45 to Virtual Sends E	25513	EXP Input 46 to Virtual Sends E	25613
EXP Input 45 to Virtual Sends F	25514	EXP Input 46 to Virtual Sends F	25614
EXP Input 45 to Virtual Sends G	25515	EXP Input 46 to Virtual Sends G	25615
EXP Input 45 to Virtual Sends H	25516	EXP Input 46 to Virtual Sends H	25616
EXP Input 47 to Virtual Sends A	257Ø9	EXP Input 48 to Virtual Sends A	258Ø9
EXP Input 47 to Virtual Sends B	2571Ø	EXP Input 48 to Virtual Sends B	2581Ø
EXP Input 47 to Virtual Sends C	25711	EXP Input 48 to Virtual Sends C	25811
EXP Input 47 to Virtual Sends D	25712	EXP Input 48 to Virtual Sends D	25812
EXP Input 47 to Virtual Sends E	25713	EXP Input 48 to Virtual Sends E	25813
EXP Input 47 to Virtual Sends F	25714	EXP Input 48 to Virtual Sends F	25814
EXP Input 47 to Virtual Sends G	25715	EXP Input 48 to Virtual Sends G	25815
EXP Input 47 to Virtual Sends H	25716	EXP Input 48 to Virtual Sends H	25816
EXP Input 49 to Virtual Sends A	259Ø9	EXP Input 50 to Virtual Sends A	26ØØ9
EXP Input 49 to Virtual Sends B	2591Ø	EXP Input 50 to Virtual Sends B	26Ø1Ø
EXP Input 49 to Virtual Sends C	25911	EXP Input 50 to Virtual Sends C	26Ø11
EXP Input 49 to Virtual Sends D	25912	EXP Input 50 to Virtual Sends D	26Ø12
EXP Input 49 to Virtual Sends E	25913	EXP Input 50 to Virtual Sends E	26Ø13
EXP Input 49 to Virtual Sends F	25914	EXP Input 50 to Virtual Sends F	26Ø14
EXP Input 49 to Virtual Sends G	25915	EXP Input 50 to Virtual Sends G	26Ø15
EXP Input 49 to Virtual Sends H	25916	EXP Input 50 to Virtual Sends H	26Ø16

③ EXP Inputs to Virtual Sends	X60	③ EXP Inputs to Virtual Sends	X60
EXP Input 51 to Virtual Sends A	261Ø9	EXP Input 52 to Virtual Sends A	262Ø9
EXP Input 51 to Virtual Sends B	2611Ø	EXP Input 52 to Virtual Sends B	2621Ø
EXP Input 51 to Virtual Sends C	26111	EXP Input 52 to Virtual Sends C	26211
EXP Input 51 to Virtual Sends D	26112	EXP Input 52 to Virtual Sends D	26212
EXP Input 51 to Virtual Sends E	26113	EXP Input 52 to Virtual Sends E	26213
EXP Input 51 to Virtual Sends F	26114	EXP Input 52 to Virtual Sends F	26214
EXP Input 51 to Virtual Sends G	26115	EXP Input 52 to Virtual Sends G	26215
EXP Input 51 to Virtual Sends H	26116	EXP Input 52 to Virtual Sends H	26216
EXP Input 53 to Virtual Sends A	263Ø9	EXP Input 54 to Virtual Sends A	264Ø9
EXP Input 53 to Virtual Sends B	2631Ø	EXP Input 54 to Virtual Sends B	2641Ø
EXP Input 53 to Virtual Sends C	26311	EXP Input 54 to Virtual Sends C	26411
EXP Input 53 to Virtual Sends D	26312	EXP Input 54 to Virtual Sends D	26412
EXP Input 53 to Virtual Sends E	26313	EXP Input 54 to Virtual Sends E	26413
EXP Input 53 to Virtual Sends F	26314	EXP Input 54 to Virtual Sends F	26414
EXP Input 53 to Virtual Sends G	26315	EXP Input 54 to Virtual Sends G	26415
EXP Input 53 to Virtual Sends H	26316	EXP Input 54 to Virtual Sends H	26416
EXP Input 55 to Virtual Sends A	265Ø9	EXP Input 56 to Virtual Sends A	266Ø9
EXP Input 55 to Virtual Sends B	2651Ø	EXP Input 56 to Virtual Sends B	2661Ø
EXP Input 55 to Virtual Sends C	26511	EXP Input 56 to Virtual Sends C	26611
EXP Input 55 to Virtual Sends D	26512	EXP Input 56 to Virtual Sends D	26612
EXP Input 55 to Virtual Sends E	26513	EXP Input 56 to Virtual Sends E	26613
EXP Input 55 to Virtual Sends F	26514	EXP Input 56 to Virtual Sends F	26614
EXP Input 55 to Virtual Sends G	26515	EXP Input 56 to Virtual Sends G	26615
EXP Input 55 to Virtual Sends H	26516	EXP Input 56 to Virtual Sends H	26616

EXP Outputs

The EXP output bus routes the inputs (①), and virtual returns (②), to the Extron EXP outputs.

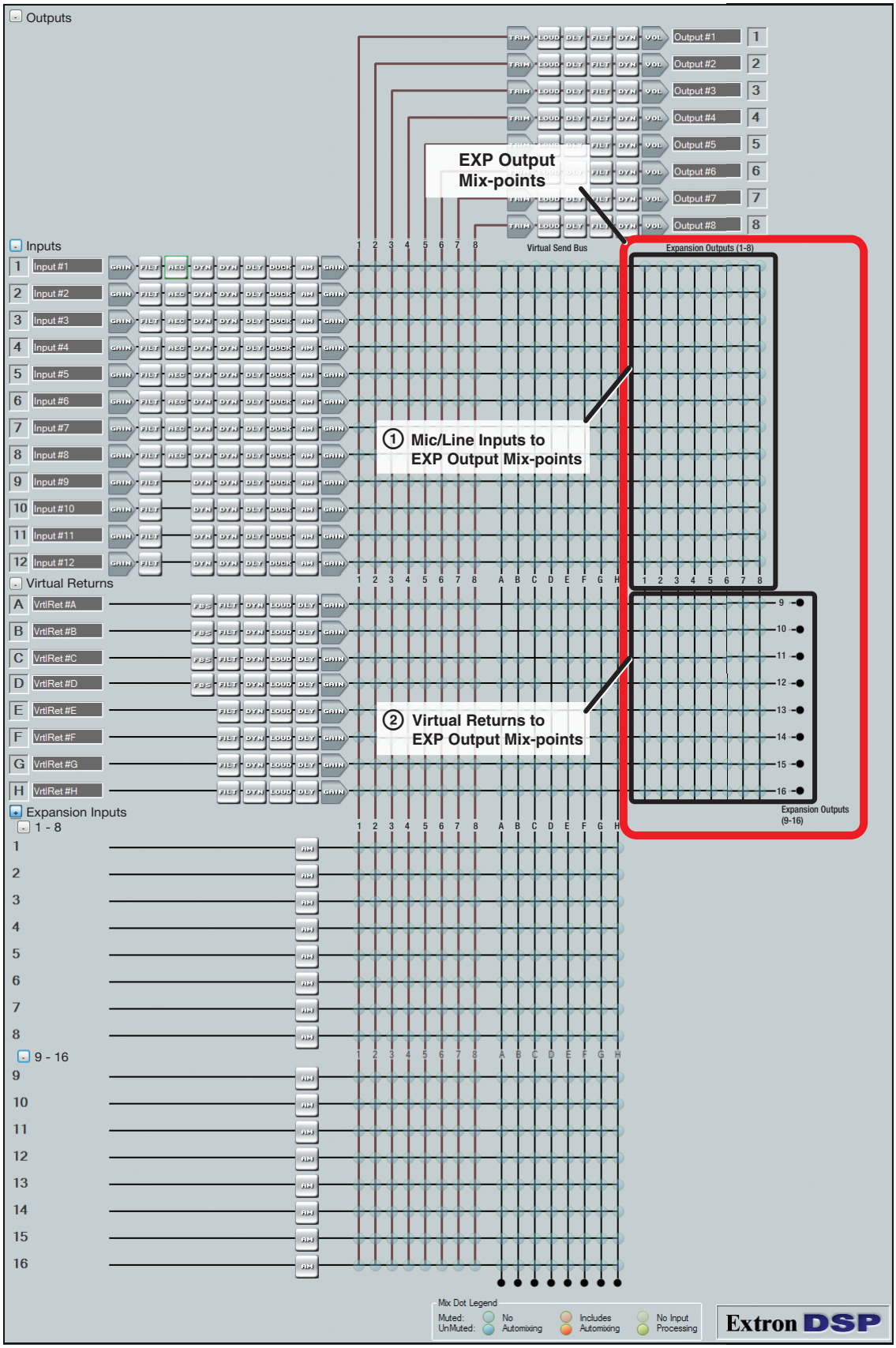


Table 9: Mic/Line Inputs to EXP Outputs

① Inputs to EXP Outputs	X60	① Inputs to EXP Outputs	X60
Input 1 to EXP Output 1	2ØØ18	Input 2 to EXP Output 1	2Ø117
Input 1 to EXP Output 2	2ØØ18	Input 2 to EXP Output 2	2Ø118
Input 1 to EXP Output 3	2ØØ19	Input 2 to EXP Output 3	2Ø119
Input 1 to EXP Output 4	2ØØ2Ø	Input 2 to EXP Output 4	2Ø12Ø
Input 1 to EXP Output 5	2ØØ21	Input 2 to EXP Output 5	2Ø121
Input 1 to EXP Output 6	2ØØ22	Input 2 to EXP Output 6	2Ø122
Input 1 to EXP Output 7	2ØØ23	Input 2 to EXP Output 7	2Ø123
Input 1 to EXP Output 8	2ØØ24	Input 2 to EXP Output 8	2Ø124
Input 3 to EXP Output 1	2Ø217	Input 4 to EXP Output 1	2Ø317
Input 3 to EXP Output 2	2Ø218	Input 4 to EXP Output 2	2Ø318
Input 3 to EXP Output 3	2Ø219	Input 4 to EXP Output 3	2Ø319
Input 3 to EXP Output 4	2Ø22Ø	Input 4 to EXP Output 4	2Ø32Ø
Input 3 to EXP Output 5	2Ø221	Input 4 to EXP Output 5	2Ø321
Input 3 to EXP Output 6	2Ø222	Input 4 to EXP Output 6	2Ø322
Input 3 to EXP Output 7	2Ø223	Input 4 to EXP Output 7	2Ø323
Input 3 to EXP Output 8	2Ø224	Input 4 to EXP Output 8	2Ø324
Input 5 to EXP Output 1	2Ø417	Input 6 to EXP Output 1	2Ø517
Input 5 to EXP Output 2	2Ø418	Input 6 to EXP Output 2	2Ø518
Input 5 to EXP Output 3	2Ø419	Input 6 to EXP Output 3	2Ø519
Input 5 to EXP Output 4	2Ø42Ø	Input 6 to EXP Output 4	2Ø52Ø
Input 5 to EXP Output 5	2Ø421	Input 6 to EXP Output 5	2Ø521
Input 5 to EXP Output 6	2Ø422	Input 6 to EXP Output 6	2Ø522
Input 5 to EXP Output 7	2Ø423	Input 6 to EXP Output 7	2Ø523
Input 5 to EXP Output 8	2Ø424	Input 6 to EXP Output 8	2Ø524
Input 7 to EXP Output 1	2Ø617	Input 8 to EXP Output 1	2Ø717
Input 7 to EXP Output 2	2Ø618	Input 8 to EXP Output 2	2Ø718
Input 7 to EXP Output 3	2Ø619	Input 8 to EXP Output 3	2Ø719
Input 7 to EXP Output 4	2Ø62Ø	Input 8 to EXP Output 4	2Ø72Ø
Input 7 to EXP Output 5	2Ø621	Input 8 to EXP Output 5	2Ø721
Input 7 to EXP Output 6	2Ø622	Input 8 to EXP Output 6	2Ø722
Input 7 to EXP Output 7	2Ø623	Input 8 to EXP Output 7	2Ø723
Input 7 to EXP Output 8	2Ø624	Input 8 to EXP Output 8	2Ø724
Input 9 to EXP Output 1	2Ø817	Input 10 to EXP Output 1	2Ø917
Input 9 to EXP Output 2	2Ø818	Input 10 to EXP Output 2	2Ø918
Input 9 to EXP Output 3	2Ø819	Input 10 to EXP Output 3	2Ø919
Input 9 to EXP Output 4	2Ø82Ø	Input 10 to EXP Output 4	2Ø92Ø
Input 9 to EXP Output 5	2Ø821	Input 10 to EXP Output 5	2Ø921
Input 9 to EXP Output 6	2Ø822	Input 10 to EXP Output 6	2Ø922
Input 9 to EXP Output 7	2Ø823	Input 10 to EXP Output 7	2Ø923
Input 9 to EXP Output 8	2Ø824	Input 10 to EXP Output 8	2Ø924

① Inputs to EXP Outputs	X60	① Inputs to EXP Outputs	X60
Input 11 to EXP Output 1	21Ø17	Input 12 to EXP Output 1	21117
Input 11 to EXP Output 2	21Ø18	Input 12 to EXP Output 2	21118
Input 11 to EXP Output 3	21Ø19	Input 12 to EXP Output 3	21119
Input 11 to EXP Output 4	21Ø2Ø	Input 12 to EXP Output 4	2112Ø
Input 11 to EXP Output 5	21Ø21	Input 12 to EXP Output 5	21121
Input 11 to EXP Output 6	21Ø22	Input 12 to EXP Output 6	21122
Input 11 to EXP Output 7	21Ø23	Input 12 to EXP Output 7	21123
Input 11 to EXP Output 8	21Ø24	Input 12 to EXP Output 8	21124

Telephone Rx to EXP Output 1	21217
Telephone Rx to EXP Output 2	21218
Telephone Rx to EXP Output 3	21219
Telephone Rx to EXP Output 4	2122Ø
Telephone Rx to EXP Output 5	21221
Telephone Rx to EXP Output 6	21222
Telephone Rx to EXP Output 7	21223
Telephone Rx to EXP Output 8	21224

Table 10: Virtual Returns to EXP Outputs

② Virtual Returns to EXP Outputs	X60	② Virtual Returns to EXP Outputs	X60
Virtual Return A to EXP Output 1	21317	Virtual Return B to EXP Output 1	21417
Virtual Return A to EXP Output 2	21318	Virtual Return B to EXP Output 2	21418
Virtual Return A to EXP Output 3	21319	Virtual Return B to EXP Output 3	21419
Virtual Return A to EXP Output 4	2132Ø	Virtual Return B to EXP Output 4	2142Ø
Virtual Return A to EXP Output 5	21321	Virtual Return B to EXP Output 5	21421
Virtual Return A to EXP Output 6	21322	Virtual Return B to EXP Output 6	21422
Virtual Return A to EXP Output 7	21323	Virtual Return B to EXP Output 7	21423
Virtual Return A to EXP Output 8	21324	Virtual Return B to EXP Output 8	21424
Virtual Return C to EXP Output 1	21517	Virtual Return D to EXP Output 1	21617
Virtual Return C to EXP Output 2	21518	Virtual Return D to EXP Output 2	21618
Virtual Return C to EXP Output 3	21519	Virtual Return D to EXP Output 3	21619
Virtual Return C to EXP Output 4	2152Ø	Virtual Return D to EXP Output 4	2162Ø
Virtual Return C to EXP Output 5	21521	Virtual Return D to EXP Output 5	21621
Virtual Return C to EXP Output 6	21522	Virtual Return D to EXP Output 6	21622
Virtual Return C to EXP Output 7	21523	Virtual Return D to EXP Output 7	21623
Virtual Return C to EXP Output 8	21524	Virtual Return D to EXP Output 8	21624
Virtual Return E to EXP Output 1	21717	Virtual Return F to EXP Output 1	21817
Virtual Return E to EXP Output 2	21718	Virtual Return F to EXP Output 2	21818
Virtual Return E to EXP Output 3	21719	Virtual Return F to EXP Output 3	21819
Virtual Return E to EXP Output 4	2172Ø	Virtual Return F to EXP Output 4	2182Ø
Virtual Return E to EXP Output 5	21721	Virtual Return F to EXP Output 5	21821
Virtual Return E to EXP Output 6	21722	Virtual Return F to EXP Output 6	21822
Virtual Return E to EXP Output 7	21723	Virtual Return F to EXP Output 7	21823
Virtual Return E to EXP Output 8	21724	Virtual Return F to EXP Output 8	21824
Virtual Return G to EXP Output 1	21917	Virtual Return H to EXP Output 1	22Ø17
Virtual Return G to EXP Output 2	21918	Virtual Return H to EXP Output 2	22Ø18
Virtual Return G to EXP Output 3	21919	Virtual Return H to EXP Output 3	22Ø19
Virtual Return G to EXP Output 4	2192Ø	Virtual Return H to EXP Output 4	22Ø2Ø
Virtual Return G to EXP Output 5	21921	Virtual Return H to EXP Output 5	22Ø21
Virtual Return G to EXP Output 6	21922	Virtual Return H to EXP Output 6	22Ø22
Virtual Return G to EXP Output 7	21923	Virtual Return H to EXP Output 7	22Ø23
Virtual Return G to EXP Output 8	21924	Virtual Return H to EXP Output 8	22Ø24

Extron Warranty

Extron Electronics warrants this product against defects in materials and workmanship for a period of three years from the date of purchase. In the event of malfunction during the warranty period attributable directly to faulty workmanship and/or materials, Extron Electronics will, at its option, repair or replace said products or components, to whatever extent it shall deem necessary to restore said product to proper operating condition, provided that it is returned within the warranty period, with proof of purchase and description of malfunction to:

USA, Canada, South America, and Central America:
Extron Electronics
1230 South Lewis Street
Anaheim, CA 92805
U.S.A.

Japan:
Extron Electronics, Japan
Kyodo Building, 16 Ichibancho
Chiyoda-ku, Tokyo 102-0082
Japan

Europe and Africa:
Extron Europe
Hanzeboulevard 10
3825 PH Amersfoort
The Netherlands

China:
Extron China
686 Ronghua Road
Songjiang District
Shanghai 201611
China

Asia:
Extron Asia Pte Ltd
135 Joo Seng Road, #04-01
PM Industrial Bldg.
Singapore 368363
Singapore

Middle East:
Extron Middle East
Dubai Airport Free Zone
F12, PO Box 293666
United Arab Emirates, Dubai

This Limited Warranty does not apply if the fault has been caused by misuse, improper handling care, electrical or mechanical abuse, abnormal operating conditions, or if modifications were made to the product that were not authorized by Extron.

NOTE: If a product is defective, please call Extron and ask for an Application Engineer to receive an RA (Return Authorization) number. This will begin the repair process.

USA: 714.491.1500 or 800.633.9876
Asia: 65.6383.4400
Europe: 31.33.453.4040
Japan: 81.3.3511.7655

Units must be returned insured, with shipping charges prepaid. If not insured, you assume the risk of loss or damage during shipment. Returned units must include the serial number and a description of the problem, as well as the name of the person to contact in case there are any questions.

Extron Electronics makes no further warranties either expressed or implied with respect to the product and its quality, performance, merchantability, or fitness for any particular use. In no event will Extron Electronics be liable for direct, indirect, or consequential damages resulting from any defect in this product even if Extron Electronics has been advised of such damage.

Please note that laws vary from state to state and country to country, and that some provisions of this warranty may not apply to you.

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